THE SYSTEMATIC MIXING GUIDE

BY ERMIN HAMIDOVIC

THE PRACTICAL HANDBOOK TO MAKE YOUR TRACKS **SLAM**

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Acknowledgements

I would like to start out by thanking James Bogdanis for his tireless work in taking this guide in its rough, verbose, infantile state and turning it into the product which you now read. Where I had been sapped and lost the motivation to go on, he took the torch and saw the guide to completion.

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Foreword

Thank you for purchasing the Systematic Mixing Guide. The guide is an independent release, and as such, every sale counts. Your honesty and integrity, in light of supporting the many years of research and accumulated experience which went into authoring this text is greatly appreciated.

The forerunner to this guide originally appeared in its first iteration on an online forum. It was intended to provide a quick reference to newer members of the engineering community regarding common mix processes in the pop, rock and metal genres. As its scope expanded to detail ever more (sometimes indirectly) relevant information, it became obvious that it would need to be contained within something more holistic and structured. This book is the final result.

It should be said outright that there is no universal philosophy to mixing. The basic gist of the process is to take a set of individual tracks and make them work in concert with each other. Everything beyond that is almost entirely subjective.

This book details my personal approach to mixing - a mindset that I've tried to develop and refine in a way that, at least to me, is as rational and methodical as possible. My hope is to give you a kick start on your mixing journey, and overcome some of the many hurdles I struggled with at the beginning, hopefully in a single, large bound.

My firm belief is that getting the right knowledge comes from knowing to ask the right questions. This book strives to detail and elucidate upon some of the main concepts behind mixing, as well as showing practical approaches to real situations, which should serve as solid starting points for your foundation as a mix engineer. Above all, I wanted to ensure that you were provided concrete, practical information regarding most topics, rather than just arbitrary ideas and theoretical dissertations.

Having said all that, this book is not one for outright newcomers to the audio realm. It assumes some degree of understanding and competency with basic audio tools and principles. If you don't feel comfortable at the very least operating tools such as DAWs, equalizers, compressors, reverbs, delays, or understand concepts such as phase, frequency and decibels, then it may be worth consolidating this knowledge before reading on.

After putting down (or digitally closing, as it may be) this book, I would hope that you would be more well equipped to deal with any sessions thrown your way and know where to look down the track for further experience and knowledge as your career continues to develop.

Happy reading,

Ermin Hamidovic
Owner, Engineer & Producer - Systematic Productions

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Chapter 1: Preparation is half the battle

Whole-mix philosophy

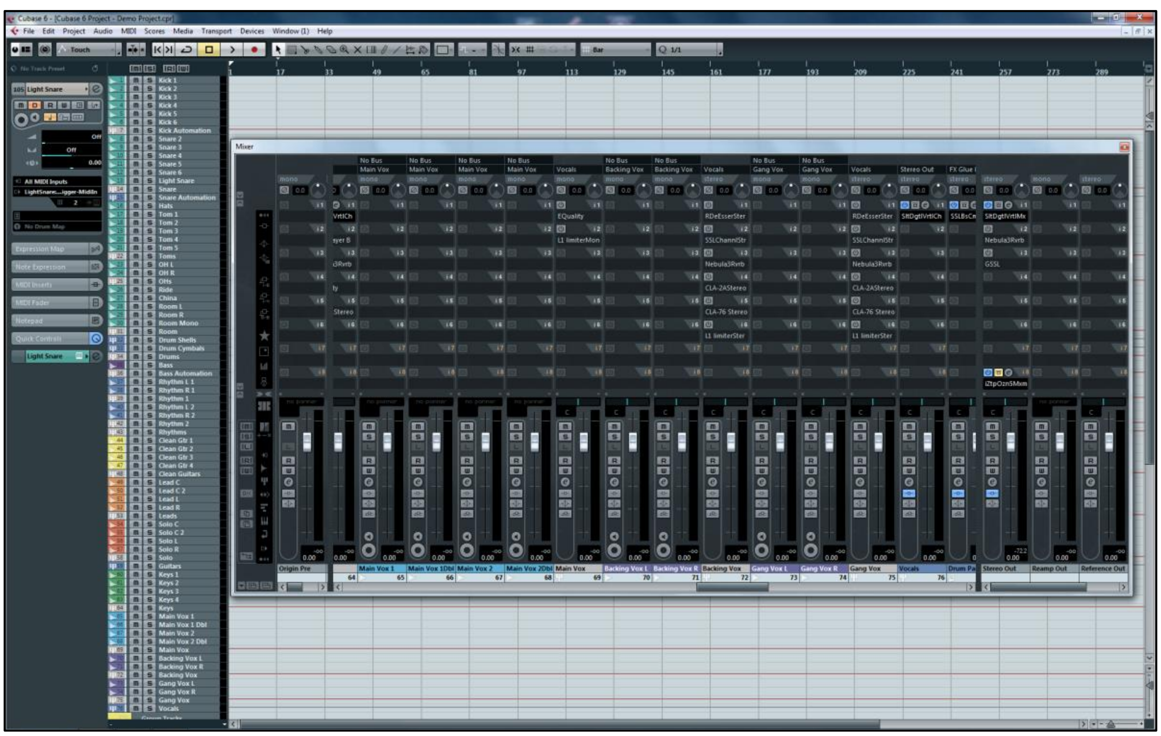
It can be overwhelming, that moment where you first line up all the constituent tracks of a new project within your DAW, or your console. It screams of potential. There is so much to be done, usually with so little time and no objective or structured way with which to tackle it. It begins to take shape within your mind - ideally, that end product that you haven't even started working toward already exists up there. The imagination to hear that mix, as you intend it to be, before you tweak a single knob is a crucial thing. You've discussed the desired direction for the mix with the artist. You already know within your mind what it should sound like at the end. So where do you begin with it all?

It starts with one word.

Workflow

Whether you use a DAW, console or anything in between, developing a workflow that matches your needs is one of the most important tools to help you create a lasting career as a mix engineer. As a DAW user, this might mean creating template sessions with pre-routed tracks and busses - with pre-loaded go-to plug-ins. It might mean creating macros to expedite faster working. As a console user, it might mean delegating certain channels to always take certain types of tracks. It might mean pre-patching your outboard to insert points where you would commonly use them, despite the nature of the project.

Whatever the case, if you hope to repeatedly work on mix projects, you will not enjoy the task of having to set everything up from zero each and every time, especially when a



majority of your projects will likely require similar routing, track layouts and processing. No, this *doesn't* mean that you will use the same processing presets for each unique project, but it *does* mean that your go-to tools will always be within reach where you need them, when you need them. So, provided everything is rearing to go and within reach, how do we start this rodeo?

Gain staging

These two words mean next to nothing to most of those brought up in the digital era of audio processing. Yet they are still the two words chiefly representative of the most important aspect of preparing a project for mixing.

You need to specify a consistent gain range for where you intend to mix. This is much easier to do in analogue environments, where everything is dictated by nominal operating levels and displayed by VU meters. It can be a whole other ball game in the digital realm, where the dB levels are somewhat arbitrary in relation to the real world.

There is currently no nominal level standard in digital workstations. Most plug-ins are configured to work around different operating levels, and most DAWs meter in dB Full Scale, which doesn't consider headroom - it simply just chops audio past the 0dB point (in a none-too-pleasing way as well!).

More and more digital mixing engineers are adopting the old analogue workflow when setting up sessions in a DAW. The general notion is that a good quality, professional analogue mixing console has around 18dB of headroom beyond the nominal operating level. Consequently, many are adjusting their individual tracks to sit around -18dB on their digital meters, with the space above reserved for transient content and dynamics. This allows an appropriate amount of headroom to mix just about any style of music, as dynamically as one might need. I would recommend following suit. Furthermore, calibrating your AD/DA converters to this reference level for using outboard gear would be wise.

Adjust the gain using the gain controls/virtual trim pots on your DAW channel mixers (or trim plug-ins, in absence of inbuilt channel gain controls). The reason for doing this is that you want the level to be pre-fader, so that all your inserts (whether they be plug-ins or hardware) benefit from the increased headroom. This also frees up your faders to sit closer to 0, giving you a better visual cue of what you're doing when mixing.

Now that our projects are all pre-routed and gain staged, we can move on to the fun stuff. But before we do, we need to visit one very important element in the mixing equation!

Monitoring

As a mix engineer, it's crucial that you can make appropriate judgment calls when presented with output from your monitors. In an ideal world, the interaction would be directly between the source material and the engineer's ears, but unfortunately this is never the case. There is always a monitoring system and physical environment in the way (at least until we work out a way to mix in an infinite room!).

One of the greatest problems younger engineers, with poor monitoring systems and environments face, is the question of whether an audible problem with a mix is actually a problem with the mix, or whether it's a problem with the monitoring system or environment itself. Half the battle is not only fighting the mix, but also the environment that creates a veil between us and that mix.

Here are a couple of methods to circumvent this problem:

1. Invest in a good monitoring system! This is crucial, and constitutes the most basic need to form your work environment. Without the ability to hear accurate detail, you cannot mix with accuracy.

2. Invest in acoustic treatment! At a certain price point, acoustic treatment becomes much more important than the monitoring system in a room. Assuming we're starting with a modestly decent playback system, we will have much more to gain by treating room flutter, modal and other coloration problems than simply spending more on monitors.

3. Reference as much as possible, and then some! The best way to learn the idiosyncrasies of your unique work environment is to listen to top-class mixes in it. Once you have a feel for how records 'should' sound in there, you will get a better feel for how to take your own mixes there naturally.

4. Listen to your work on other systems.

5. Balance low-end on headphones. Since no room is perfect, this can help an absolute ton under most circumstances – provided the headphones are of quality.

6. Invest in professional mastering. Mastering can help, within reason, to work through and attenuate mix flaws relating to poor working environments. Beyond this, you can cultivate feedback from the mastering engineer and take into account what to concentrate on improving next time.

The fine art of *listening*

Any single action taken during a mix should be preceded by active listening. This is after all, what mixing is all about. Everything comes down to how well you hear the source, the judgment calls you make regarding the source, and how you put your tools into effect to mold the source into your vision of the whole.

Many younger and even seasoned engineers claim that they have trouble dialing equalizers and compressors - the bread and butter of mixing tools! If we were building a house, they would be our bricks and mortar. You certainly wouldn't want a builder with no knowledge of brick-laying to build your house (unless you like to live on the edge, quite literally), so you most certainly don't want to be the kind of engineer who fumbles their way through dialing the most essential of mix tools.

The solution is deceptively simple. You simply need to learn how to *listen.* If you can isolate the problem, and construct a solution, then the task of picking an appropriate tool and dialing it to purpose becomes amazingly intuitive.

Equalization

If we are after a 'clean' mix, then it helps to think of equalization surgically. The idea is that you will listen to the source, whether it be a drum, piano, vocal or whatever else and immediately begin to isolate 'problem' zones. You might find there are particular resonances and artifacts within the frequency response of the source, and your mind should gravitate toward them. If it doesn't at first, don't worry. After a period of going through the process of surgical equalization, your mind will isolate problems automatically without much active intervention on your part (yes, to the point of being maddening when you're forced to endure listening to poor mixes!). So, how do you know they are actually problems? Well, because when you address them the source will immediately sound more open, balanced, and willing to work as part of a team.

Standard equalization practice is to isolate in your mind, as best as you can, where the problem zones lie. Once you've done this, you can go in with the appropriate equalizer and jog a tight peak boost around the zone where you felt the problem was. This has a two-fold effect. The first is that you will isolate specifically where the problem lies. The second is that you will have honed your hearing marginally and related it to real-world numbers. This will continue to help you throughout your career. Now that you have isolated a problem zone with your boost, you can then turn it into a cut. Cut only as much as you need to. The best way to find the sweet spot is by jogging the cut up and down, and also altering the center frequency left and right, until you can hear the artifact disappear. Practice this often, as it will become one of your most used techniques on virtually every project you ever do.

Consider what you could remove before you begin adding. Our goal is to 'clean up' the source so that it will be ready to work in concert with the rest of the mix. If it's lacking a bit of body or sheen, then that can be easily fixed with a nice colorful boost or two, but the subjugation of unpleasant oddities and artifacts should come first.

At this stage, it's worth dispensing with the myth that 'equalization should only ever be done with the full mix running!' The truth is that when you have several resonant overlaps and dozens of clashes all over the frequency spectrum from dozens, if not hundreds of unprocessed tracks running in concert, it's hard to know up from down. Simplify. Make it easy on yourself, and start your journey by soloing the element you are working on at this part of the process. The true power behind attaining perspective is not exclusively in using EQs when the full mix is running, but knowing when to EQ in solo as well as EQ'ing to balance something as part of a whole.

Contrary to popular thinking, the most powerful feature of your favorite equalizer is in fact the 'bypass' button. This ingenious feature gives you the ability to ensure that your processing has benefitted, rather than diminished the signal in a single press. It also encourages you to practice correct gain-staging, as the perceived volume of the processed signal relative to the unprocessed signal should be about the same. Use this feature regularly, as it will reveal the artifacts you've subdued in the original signal and ensure you haven't followed the processing rabbit hole too far down.

When all is said and done, ensure that you listen to the track in context with everything else. Always try to maintain a global perspective rather than getting carried away with microscopic details. Cleaning up a track with surgical EQ is all good and well, but it won't help you if that track doesn't glue with anything else in the mix.

Compression

Compression is one of the most misunderstood and often misused techniques in the entire realm of audio engineering. Many use it simply because they feel they have to, with no particular regard or knowledge for what they are doing to the signal. This is an area to tread carefully because as well as being one of our most powerful tools, the compressor can be one of the most prone to destroying our mixes outright. I'm sure there's a Spiderman quote about great responsibility to bring that concept home, but

let's stay on topic instead.

Compressors are, in essence, dynamic range reduction tools. Hence the name - they 'compress' the signal. They shrink the dynamic footprint of the source so that it becomes conducive to working within an arrangement. This allows us greater control over the level and dynamics of our mixes and ultimately allows for finer sculpting.

As well as controlling dynamic range, most compressors also allow us the ability to shape the envelope of the signal we are compressing. This is where the common 'Attack' and 'Release' parameters come into play. The attack effectively determines how long it takes a compressor, after the input has exceeded the threshold, to reach a set percentage of its final gain reduction value. The release works in opposite, determining how long it takes a compressor, after the input has dropped below the threshold, to reach a set percentage of its original gain value. This makes compressors great tone-shaping tools. They can help to affect how far forward or back in a mix the source will sit, how much it will breathe, how small its footprint will be in the mix and so forth. They help shape how quickly, and how aggressively the compressor acts upon the signal.

Now, you're not reading this because you want a 101 on the basic functions of your garden-variety compressor. You want to know how to dial the thing, right? Allow me to first bore you with the self-indulgent, esoteric diatribe that most in my position feel tempted to unleash upon newcomers before actually moving onto the practically relevant information you actually want. Dialing compressors is somewhat of an art form, and it can take not only years, but decades to truly refine. Combining the inherently unique characteristics of different types of compressors with the permutable ways of dialing and sequencing them gives an astonishing amount of possibilities. You can rest assured that many more of those will sound 'wrong' than they do 'right'. Picking which compressor to use on which source can at times be half the battle, so you will need to familiarize yourself with as many types of compressors as you can, and their respective tonal characteristics.

To start with, when listening to a signal prior to compression, you need to focus on the dynamic qualities of the signal. Work out what you can do with compression, and what's best done via other avenues, such as volume automation, re-tracking, sampling etc. It helps to form an idea of how you want this element to sit within your mix - it will determine how you will ultimately dial your compression chain. Assuming we are using a garden-variety compressor with simple Threshold, Ratio, Attack, Release and Output Gain controls, we can proceed as such:

Dial up the Ratio fairly high. Anywhere from 8:1 on up should do. Set the Release & Attack to the fastest they will go. Lower the threshold until you're compressing the signal quite heavily. What you have here is a compressor that's acting much like a limiter. It's not giving the signal very far to go, and with most compressors what you're hearing now should sound god awfully bad. Good? Ok, let's move on.

Start by opening up the Attack. This will determine the size of transients you allow through the compressor. If we're compressing drums, you will generally find that you

prefer slower attacks on direct tracks, when searching for punch. On vocals you might find anything slower than a quick/medium attack lets too many plosives and other distractions through. Essentially, the faster your attack, the more you stifle the transient content of the signal. And the slower the attack, the more natural, albeit less controlled it will sound.

Next you will want to set your release. The release is a fairly straightforward one as you want to ensure that your compressor has mostly released by the time the next phrase, drum hit, or whatever else has come into action. The shorter your release, the more you emphasize the resonant, sustained qualities of whatever it is that you're compressing. The slower your release, the less of that sustained quality will jump up after the compression, and the more it will start to eat into the next transient. If your release is too short, you may emphasize pumping qualities in the signal, but if it's too long you may eat into too much coming transient content. A good example to use here is to assume we're dealing with a quick burst of double-bass drums. Most who have dealt with double-kick parts will know that they tend to jump out of the mix quite profoundly and need to be dealt with. Let's say that for some reason volume automation isn't an option and we'd rather our compressor deal with them. In this case what you would want to do is set the compressor so that it releases normally for the sparse single-kick parts, but come the double-kick stuff, the release will still be going through its motion by the time all the successive hits have come around. The effect here? The first kick hit (the down beat, which is helpful to emphasize) will be relatively normal, but every successive kick in the pattern will be ducked down in volume, and won't jump out as much as it would otherwise. Presto! Volume automation with a release control.

After this you will want to balance your threshold and ratio controls until you find a happy balance for whatever it is that you're compressing. A higher ratio will get you closer to limiting and will have a much more pronounced and aggressive sound. A lower ratio will be more gentle and transparent in nature, allowing you to get away with compressing more of the signal, albeit to a lesser degree. The rest is simply down to practice. The more you compress and the more you mix, the better feel you will develop for your compressors and what they are capable of. Above all, you will learn what to avoid doing, which is possibly the most important lesson of them all.

Naturally, you won't always dial compressors in this manner. This was simply a case in point, where hearing a compressor working at extremes can help to illustrate what it's actually doing to the signal, and how to best shape it. With experience under your belt, you will find that each of your compressors will have certain sweet spots, and you will be able to dial to those intuitively over time, much as you would dial a tone on a guitar amp. Rather than using one type of compressor for everything, and varying the settings considerably to suit, you will start to reach for different types of compressors, with sweet spots more attuned to what you are compressing from the outset. All of this is a factor of experience and time, and will develop in concert with your individual mixing personality.

Compressor basic controls

Threshold: The point beyond which the compressor will start to act.

Attack: The amount of time it takes the compressor, after a threshold overshoot, to arrive at a set percentage of its final gain reduction value.

Envelope/Knee: Determines how gradually a compressor comes into action. A soft knee will mean the compressor will slide into action gradually before a threshold is exceeded. A hard knee will mean the compressor will jump into action suddenly after exceeding the threshold.

Ratio: The degree of the compressor's action upon the signal, expressed as a dB ratio between the input and output signal levels.

Release: The amount of time it takes a compressor, after a threshold undershoot, to arrive at a set percentage of its original gain value.

Output Gain: Simple make-up gain control to compensate for the level lost when compressing.

Basic Compressor Topologies

VCA (Voltage Controlled Amplifier): Generally seen in fast, low-distortion compressor designs. Can be tight and snappy. Good all-rounder - particularly good on drums.

FET (Field Effect Transistor): Generally round and fat sounding. Can have pleasant distortion characteristics under heavy gain reduction. Good on vocals, bass, synths & drum rooms/busses.

Opto (Optical Light Cell): Generally natural and gentle sounding. Smooth compression characteristics. Good on acoustic instruments and vocals.

Spectrum legend

Throughout this book you will find terms like ‘sub-bass’, ‘mid-bass’, ‘low-mid’, ‘core-mid’ etc. being thrown around. In order to avoid confusion, as the definitions for these terms are arbitrary, we’ll lay out a legend here that generally approximates the band of frequencies being talked about by each of these throughout the book.

Note that these are all approximate, and may vary a tiny amount from example to example, depending on context.

Sub-bass:	60Hz and below
Bass:	60 to 120Hz
Mid-bass:	120 to 250Hz
Low-mid:	250-600Hz
Core-mid:	~1kHz
High-mid:	2kHz to 5kHz
Highs:	5kHz to 10kHz
Super-Highs:	10kHz+

Saturation

In the yester-years of mixing, saturation was an everyday part of work. Consoles, by nature, had a lot of circuitry, and transformers were used quite frequently as a common way of designing equipment. They inherently colored anything that passed through them. This coloration was seen as taking away from the purity of the performance. Equipment designers worked for decades to improve the distortion figures on their gear, constantly searching for a more transparent sound.

We live in the digital age, which has gradually overtaken the audio industry in many aspects. Early digital recording mediums and converters were, at best, less-than-ideal sounding. However, once digital truly began to entrench itself in the new millennium, the working methods consolidated and the equipment became usable, we started to find ourselves in an ironic situation.

Our recordings were now 'too clean'. The lack of saturation anywhere in the recording path, sans the microphone preamp itself, caused a certain pristine, tight and sterile aesthetic to form around recordings. Many engineers found these to not sound

euphonic, as despite their great fidelity, the instruments sounded disconnected, thin and somewhat fake, despite the fact that they were truer to the recording.

We have taken a step back, hybridized our workflow and attempted to take all the best parts from the colored gear of the past, and compliment it with our new, transparent recording equipment.

Microphone preamp saturation

This is a great one for any midrange centric elements like drums, guitars or vocals. Turning a high-quality microphone preamp's gain into the 'red' zone can give a grit to the signal, which helps it survive dense mixes.

If you don't have a high quality preamp with which to saturate or distort your tracks, then don't fear. There are a lot of capable preamp distortion plug-ins around which can get the job done. Simply find the one that works for you, and experiment with how it can improve your tracks. Go easy, as overuse here can make things sound quite harsh.

Console saturation

The way this relates to mixing, specifically, is the realization that consoles of the past certainly had a 'sound' that they imprinted on the records that were mixed through them. This sound usually involved a sense of togetherness and glue, which was not possible to get using standard processing techniques within a digital environment - at least not until recently.

There are a number of console saturation plug-ins out at present and many more coming down the track. In countless tests the simple 'on' sound of the plug-ins emulating the desks was found more euphonic than the clean, disconnected sound of straight digital mixes. Simply having these plug-ins present is enough to let a mix come together faster and sound more cohesive in the end. These can be loaded into template sessions in a DAW, and set up along with the standard gain structuring of a mix session. Mixing 'into' the plug-ins will ideally help you get where you're going faster, and make it sound better in the end to boot.

This form of console saturation is usually very light and transparent. The benefit comes when it's stacked across all the tracks in a project and creates a subtle sense of depth. It's a good starting point, because it's very hard to go wrong with.

Tape saturation

Tape saturation is an area where digital is still struggling. Despite its best attempts, in order to capture all the nuances and nonlinearities of magnetic tape and its machines takes a gargantuan amount of CPU power and is a hard thing to achieve accurately.

Having said that, there are saturation plug-ins around, which take the idea of the beneficial qualities that tape imprints and apply them in a simpler model. These may be

as simple as slight saturation and a dynamic EQ curve, but even so can be enough to help a mix along.

At present it may not be a good idea to put a digital tape simulator across the master bus of an entire mix, but it can certainly help when put on acoustic instrument busses, drums, vocals and whatever else you find they help with.

As it stands, if you want the sound of tape, then you need to use tape. If you simply want gluing saturation to help your mix along, then delve into the many, many tape saturation plug-ins on offer.

Chapter 2: In Soviet Russia, drums slam you!

Drums are one of my personal favorite things to mix. They tend to form the very extremes of a mix, from the cymbals - which are generally the highest and airiest of elements - down to the kick, which in many cases is the lowest. As such, they become the framework of a good mix and thus getting good drum balance is critical.

In the modern era, as many people would be used to dealing solely with sampled drums, as there are those who deal with raw drum kits. The principles when working with both do vary slightly, but at its heart the goal is always the same. You want to find a sound that suits the music.

For the purpose of this tutorial I'm going to focus largely on my own approach and the one I feel creates the best balance of vibe and punch - augmenting real drum kits with samples. This means there will be a large focus on keeping as much of the raw drums as possible and/or desirable. The principles will be equally applicable, albeit less intensive, to those dealing solely with sampled drum kits.

Skimming over tracking

Drum recording could be a whole bible unto itself (and actually is, if Glenn Fricker has anything to say about it!) so we will only briefly touch on the ideas behind tracking drums, and instead spend the majority of our time talking about how to mix them into a record.

If your intention is to keep as much of the raw kit sound in your final mix as possible, it goes without saying that you need to get the fundamentals right. Unfortunately for you, drums are arguably one of the most difficult elements to get right across the board when tracking.

Here is a quick checklist of some of the things you need to get right in order to succeed in keeping a quality raw kit sound on your record:

- ✓ Very good player, who can hit consistently, and consistently hard
- ✓ Very good drum kit, which sounds open and minimizes muddy resonances
- ✓ Very good cymbals, which have minimal harsh resonances, and minimize their crossover with the vocal range (so, not big and low pitched!)
- ✓ Brand new heads, ideally switched after every song
- ✓ A knowledge of how to tune those heads optimally (much harder and more creative than tuning guitars)
- ✓ Very good room. For drums, the room tone is crucial, as it becomes a significant part of the drum sound in our mix
- ✓ Minimal spill in your close-mics
- ✓ An array of great quality microphones
- ✓ An array of great quality microphone preamps
- ✓ Good quality, 16-channel AD/DA converter

So… intimidated yet?

Recording quality drums costs money - a lot of it. That's just the reality of something which has so many point sources that need to be isolated and treated individually to each other on a record.

It's no wonder that more and more projects are skimping on drum tracking, and instead using samples to drive the record. Many are even completely circumventing real drum kits, and using entirely electronic drums.

Now, in many ways we can circumvent those above requirements, depending on the type of project we are working on, and the type of drum sound we are shooting for.

Modern metal has the advantage of currently sporting an aesthetic which favors very direct drum sounds, with minimal room and acoustic spill. In many ways, this allows us to circumvent our need for a great room. In fact, if you are able to record drums in a decently dead room, whether it be small or otherwise, you can then isolate and process your drums with more precision and simply apply reverb, or use room samples (we'll get to that) in order to create an artificial sense of space. This gives you a greater sense of control and for many ultra-modern sounding metal productions is actually favorable over using a large, live drum room and acoustic ambiance.

Rock music on the other hand, can be a little more reliant on natural acoustics. A quality drum room is desirable, even if simply to retain the tightness of the bottom end on room and overhead mic tracks, as they are often used to solidify and enlarge the size of the drum sound.

Getting to 'Point A'

There is a lot involved in preparing drums for mixing. The tracking and editing phases are usually substantial. Much of an engineer's personality can be found in how they tackle drums. As such, I can only hope to give a rough outline of common procedures and what you may or may not wish to do with drums. For the purpose of this guide I'll assume that we at least have an acceptable raw drum sound to work with.

Phase

The very first thing you'll want to do after the drums on a project are all tracked is pull the DAW session up and start checking phase. Good phase is critical to raw drum sounds and can be the difference to something that stays thin and washy, no matter how much you process, to something which is dense and solid. There are two major ways to go about correcting phase after a recording.

The old-school way is to simply sit there, listen to the overheads, switching the close-mics in and out, and inverting the phase on the channels, listening to hear which sounds 'larger'. The dead-give away is the setting that gives the drums more body. The more body, generally the more in-phase the drum.

So that's fairly simple, right? Well, yes, but it's also somewhat limited. If your drums happen to be 90 degrees out of phase with the overheads, then inverting the phase will do nothing for you.

How do we get around this blind spot? Well, that brings us to our second method for correcting phase, which some modern tools have made very convenient for us. Devices, such as Little Labs' IBP and plug-ins, such as Voxengo's PHA-979 allow us to physically shift the phase of the incoming signal in degrees. This flexibility allows one to place the microphones on a multi-mic'ed source where they would individually sound best, and then compensate for phase afterward. In the past, the inverse order of priority would have always held true.

Bear in mind that if you equalize using minimum-phase EQs, you will be altering the phase relationship between the drum tracks. The logical solution in cases where phase coherence is critical would be to use linear-phase EQ. Otherwise, make sure to continue checking your polarity inverse switches as you work, ensuring that your EQ moves haven't pushed anything too far out.

Time-alignment

Another faux-method of enlarging drum sounds is through artificial time alignment. Imagine you have an overhead track and a snare track. Naturally, the transient of the snare is displaced on the overhead track, being later in the time domain than on the close mic'ed source. What you can do is simply drag the overhead track forward - or the snare back - to closer align the transients. Due to the doubling up, this will perceptibly give you a stronger attack due to the hit fattening up, as well as increasing in volume.

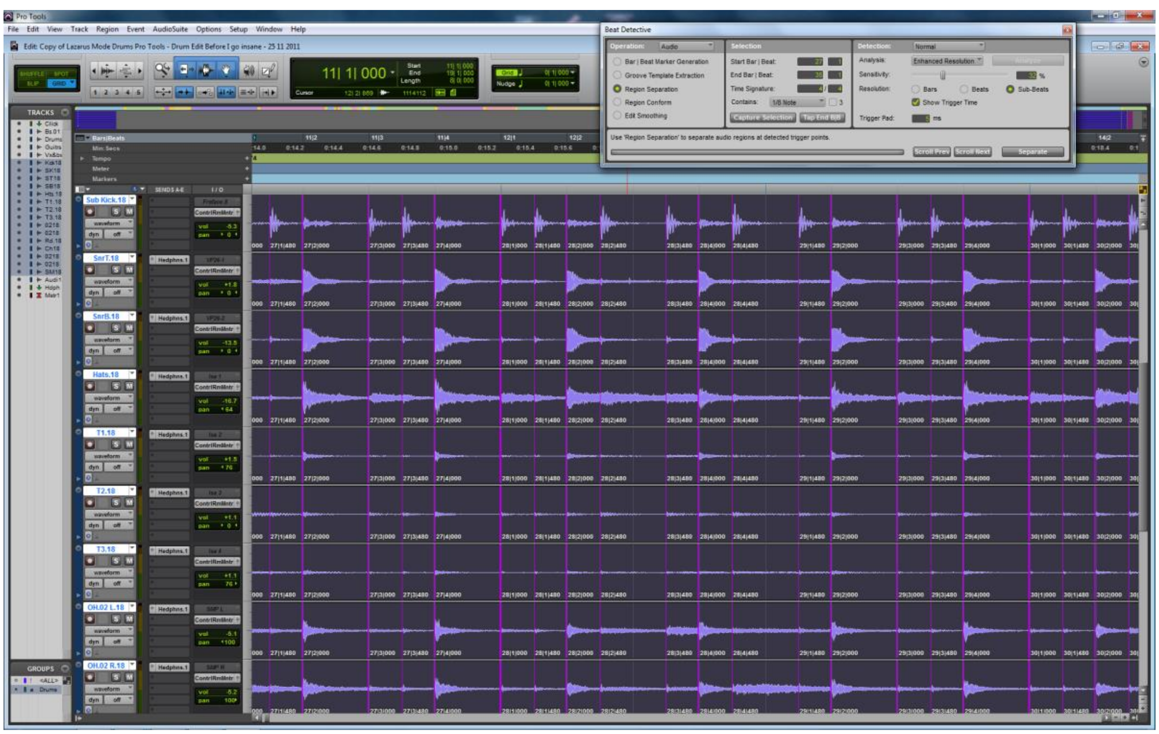
This is no replacement for adhering to the minimization of phase cancellation during tracking, nor using a dedicated phase correction tool. But, it can give you an extra

option to create the sort of drum presence you're after from the outset, without ever touching an EQ or compressor. Bear in mind that overdoing this method can create its own issues. Having no 'breathing' space between ambient tracks and close mics can make the kit initially seem more powerful, but it will also diminish the inherent sense of space within the recording.

Editing

This part is a love/hate of modern engineers. In many ways this part of the process is where the magic truly happens, as commonly average performances are tightened up into something inhuman. It's worth understanding that the vast majority of drummers out there don't have what it takes to record a world-class quality performance on drums - in metal, with its athletic demands, this is even more the case. Some do it because they aren't adept at any other instruments, or because it was the easiest way, but others – a rare few – are truly gifted and invested in their percussion. With that in mind, cast yourself back and realize that 99% of your time will constitute of working with the first kind. A lot of the time you simply need to throw the kitchen sink at the drums (if you're not tempted to include the drummer in that metaphor too) just to make it sound like a passable performance. This is relatively normal when working on modern music in this day and age, so grit your teeth, lube up and get ready to bear with me.

There are multiple ways to edit drums depending on the DAW you use, and once again, this could be a tutorial series unto itself, so we will only touch on the concepts and less the techniques.



The first order of the day, generally, is to tighten the timing of the performance. This can be done a number of ways, from slip editing, to using a Beat Detective style tool, to

using an Elastic Audio style tool.

My preferred method is the use of a Beat Detective style tool, which is capable of specifying the strength of quantization. Slip editing tends to result in sounds that are a bit too rigid for my taste. Your own may differ! However, it's worth noting that using a tool with a strength control will give you the flexibility to choose just how much of the drummer's original swing you want left in there. This can be crucial in rock music, where putting something entirely to the grid is the quickest way to kill the groove. Remember, the drums become the basis of the timing and swing of the record! If they are too rigid, then every other element laid on top is likely going to feel that way as well.

Now, assuming we've timed the performance to our liking, we can move on to manual silence editing. This is an optional process. Some people like having bleed in their close drum tracks, which is absolutely fine if they're after that aesthetic. If, however, you want extra separation and cleanliness within your mix, you will want to do this.

A common set of tracks to start with here is the toms. Since the toms are generally used as a fill element and often only come in once every few bars, the ambient spill on their tracks can be discarded. The idea is to chop the audio at the start of each transient, leave the entirety of the hit, all the resonance that you want, and chop again after that sustain has ended. You can create your own release envelopes here by using fade-outs of your desired length. Some people will also manually silence hi-hat and ride tracks during this process. I personally don't, as I tend to not mind the bleed from hat and ride tracks. In a worst-case scenario, I will simply silence an irritating ride track during the mix, either by automating the channel mute on and off or chopping and muting regions. The hat track is commonly so low in the mix that any bleed from it becomes largely inconsequential.

Now that this is done, we want to move on to placing some clean hits. Assuming that you were wise and sampled the entire drum kit at the very start of the session, you will have a collection of samples to work with, across all velocity layers.

The first order of the day is to get clean tom hits and place them at the end of every fill that has cymbals succeeding it. This way the cymbal bleed that frequently pops up when the drummer crashes on beat 1 of the following bar won't come through and mess up your balances. Use your own discretion here. Replace the hits where you see this bleed to be a problem.

After the toms are all done and consolidated you want to move on and repeat this process for weak snare and kick hits too. When this is finally finished, and your drums are all consolidated, you have yourself a ready drum track with which to begin mixing! You will thank me for being so thorough during the preparation process, as this pays off in dividends during the mix, saving you both time and hassle.

Now we're at 'Point A', so?

We're all ready to mix! What you do here and how you attempt to coax tone from the drums will depend in many ways on your personal inclination, and the sound which the song demands. Pop/rock and metal drum mixing do tend to differ in some fundamental

ways, so I will try to outline common approaches to both whenever practical.

Overheads

A common starting point is the overhead tracks. The way these are treated tends to differ fundamentally between most rock and metal music, so I will split this part here.

Metal approach

Start by jogging a high pass up the overheads until the bulk of the drum hits and their body disappears. Your general intention here is to use the overheads as cymbal microphones. Excessive drum spill is undesirable (although not catastrophic). If you find you get a lot of snare poking through, you can use a simple brick wall limiter to take the peaks down. Otherwise, you can set up a compressor that's keyed by the snare track and this will duck the overheads each time it's hit.

Pop/Rock approach

Start by setting a reasonable high pass filter. For me this tends to be anywhere from 90Hz up to 250Hz or so. A lot of this will depend on the quality of the room, which will determine the quality of low-end you'll be getting from the overheads. Don't be afraid to crank some lows into them. Try to see if you can get some beef and body from the drums here. Combining this with the direct tracks will give you more of a 'pillowy' characteristic to the low-end. Since we aren't high passing drastically here, we will likely be dealing with a fair amount of room mud. Check the area between 300 and 700Hz to see what's going on. Usually a wide scoop around there is enough to clear things up and get it breathing.

Both approaches

Now that we have the low-end and mids out of the way, we can focus on some universal principles on processing overheads. Bear in mind that the techniques listed in the above approaches aren't necessarily exclusive to their respective genres. You can do a metal record and still keep a lot of drum sound in the overheads. Feel free to be adventurous and define yourself.

The first thing to bear in mind is a potential trouble zone between 3kHz and 4kHz. This is part of the critical vocal range and believe it or not, your cymbals will eat into it. A common approach is a wide scoop around this area as a starting point. Jog the cut around with your vocals going until you find that magic spot where they start to breathe more naturally. In the meantime, make sure you haven't sucked all the life and presence out of your cymbals.

The next step you may want to try is a gentle high frequency boost. This can help give the cymbals a greater perceived sense of 'air', which will generally open up the whole

mix. Try a high-shelf between 8kHz and 10kHz. Take it easy, as things can get harsh here.

At this point, further processing will become very circumstance-dependent. Most cymbals will have resonant frequencies in a few spots and your job from here will be to tame these as much as possible, without killing all the life in the overheads. Similar principles apply here as with guitars and fizz. These are generally tight resonances, and being such they just love to eat into the clarity and front-to-back depth of your mix. You'll be amazed at how much clarity there is to be gained here once you refine the art of 'ringing out' the cymbals.

If your cymbals are sitting awkwardly after all this, seeming harsh and thin no matter what you do, try low-passing the overheads. There is no shame in this, and it can help solidify the cymbals - injecting a perception of 'body' into their tone. It goes without saying that taking this overboard will result in a fairly dull and dead-sounding mix.

Compression on overheads is a personal matter and will depend from individual to individual. I personally do like to compress overheads slightly, and I do mean *slightly*. It's more for the envelope control than control of dynamics. If you find you have a very uneven overhead track, it may very well be a better idea to automate intensively instead of compressing, as the compression will bring along with itself negative effects such as pumping, tearing cymbals, with increased resonances you may not want.

Reverb can be a great tool to aid in creating a larger sense of space around overhead tracks. One of the great things about using it here is that it effectively acts as a 'wetness' slider for the entire drum mix. If you've tracked in a small room you can really use reverb to effect, in order to enlarge the soundstage and give the impression of much larger drums than what you started out with. Not to mention you can get some increased cymbal sustain. Room, hall and plate reverbs are all valid here, depending on what the music calls for.

As always, be mindful of what does and doesn't need to be done. If you've tracked great cymbals with great mics in a great room, you may not even need to do anything more than the wide presence region scoop. So try to maintain perspective. Some rooms and types of cymbals are much more 'tracking-friendly' than others. It can be very difficult to get an open, even cymbal sound in smaller spaces, so you will inherently chase your tail with more processing in those. What you put in is what you get out, so always strive to get the quality at the source rather than the mix - this is absolutely crucial for overheads, which are very limited in their ability to take editing and augmentation after-the-fact. Reference to your favorite records if need be for guidance.

Kick

After the overheads are done, we gain our perspective - our 'whole kit', or 'cymbal' sound as it may be. With these in place we can start throwing the other drums in, one by one, and coercing them to work together.

Kick EQ

The bass drum commonly demands a particular EQ curve which is desirable for mixes.

Many times this is achieved by scooping out the region between 200Hz to 400Hz, widely, and heavily, followed by strategic low and high boosts. This will vary from kick to kick, project to project, but in general it’s a good idea to get a kick’s thump centered between 50Hz to 70Hz. This creates a good balance of low punch, whilst remaining high enough for most average stereo systems to be able to reproduce the bulk of it.

Depending on the tempo of the music, you may want to start with a high-pass filter. If you’re dealing with quicker, energetic music you can generally reject everything below ~36Hz. In many cases the content down there simply results in mud and excess headroom waste. The sub-bass content in that region is not musical and is more an element that is ‘felt’ on larger systems. So, if you’re mixing a ‘house’ track, then you may want to reconsider high-passing this drastically, as the basis of your track might end up the wimpiest sounding.

You might find under many circumstances that the kick has significant content between 100 and 200Hz. This can be dangerous as it conflicts with where your bass guitar is likely to lie. If necessary, feel free to take some energy away from this area, as it will immediately make your kick feel deeper, and your mix cleaner. As always, too much removed and you will thin the sound out too much, so be wary.

Low shelves tend to work well with kicks, unless you’re after something particularly ‘peaky’ in the low-end (in which case you may want to use a ‘peak’ boost - duh?). Depending on how sub-heavy you desire the kick to be, feel free to center your low boost anywhere from 50Hz to 80Hz as a starting point. The higher you go, the more present the kick will be on smaller systems, but the more likely it will be to conflict with your bass guitar as well. It’s all one big compromise and trade-off game, so make sure you know what type of sound you’re chasing *before* you chase it.

In regards to high-end boosts, the sweet spot tends to be between 6kHz and 10kHz for most kicks. It really depends on where you would like your high-end slap to lie, but a lot of the time the choice is self-evident as the kick will immediately sound better in the mix when you find it. If after all this you find the attack of the kick is still buried, you can try boosting within the presence region. Between 2kHz and 4kHz will lead to really present ‘clack’ type sounds, which in some ways characterize a current, very prolific, pre-processed aftermarket sample library.

Ultimately, if you find yourself boosting a lot in the high-end region, to the point where the kick is quite harsh and thin when solo, but drowned when in the mix, it may be advisable to look at the rest of your mix instead. Chances are that the guitars, overheads or synths are too bright in the presence areas and are clouding the attack of your bass

drum.

Kick compression

You generally want a slow attack, moderately fast release compressor. Anywhere from 3:1 to 6:1 in ratio tends to work well on a majority of compressors.

The idea here is to add punch to the transient of the kick, rather than dynamic control. If the kicks are very inconsistent, your better options are to either automate them or sample blend/replace. The type of compression that tends to work well for adding punch to kicks is VCA-style. SSL channel strip compressors, Distressors, DBX160VUs - all great here.

The release time on the kick compressor can at times work as a faux automation tool for you. Picture a song with a lot of double bass parts. If for some reason you don’t want to manually automate each section of double bass parts in volume/high/low-end you can simply slow the release of your kick compressor to overlap with each successive hit in the double bass run. This will have the effect of dropping kick volume during those runs, as well as somewhat dampening the transient.

A limiter on the back end can be good if you’re dealing with a largely natural kick. This can stop some of the stronger hits overshooting with massive, slappy transients. Consider setting the limiter very conservatively, as it’s almost too easy to kill the punch with one.

Kick out

If you have access to an 'outside kick' microphone, or perhaps a sub-kick, you want to use it to enhance the overall kick drum tone. Note that many popular sample packs already have this track pre-blended, so if you're dealing with one of those, this section is superfluous.

Ensure that the track is at the correct polarity by running it alongside the 'kick in' track, and using a polarity inverter all the while listening for the bottom-end energy to present itself. You may want to take this a step further by bringing the 'kick out' track forward, in order to time align it with the 'kick in' track. This will ensure a phase coherence and solidity to the low-end, which is crucial for this track to do what it needs to do.

If you are intending to use this track simply to fill out the sub energy of your kick, processing it can be as simple as running a drastic, steep low-pass down to anywhere between 100Hz to 200Hz. This way the track becomes a very basic 'low-end' slider for your mix, ensuring you can pump more energy into the lows without needing to resort so much to phase-shifting EQ on your dominant 'kick in' track.

If, however, you are using this track to augment the midrange and attack character of your overall kick, then you can treat it similarly with EQ as you do your 'kick in' track. If you're looking to avoid complications, simply bus this track together with the 'kick in', and make all your significant equalization moves there. This will ensure phase coherence between the two, regardless of the severity of your EQ moves.

Assuming you would like to cook this track a little before combining it with the 'kick in',

then consider these areas. 200Hz to 800Hz will almost undoubtedly have some serious junk build-up which you will need to attenuate with a wide, aggressive scoop. Sub content below ~36Hz will once again likely need filtering. You can choose to boost some content in the 2 to 4kHz region in order to augment the attack of the kick, but be very mindful of the spill you're bound to bring up, as this microphone is outside and prone to picking up the rest of the kit. Finally, consider leaving any content above 4kHz for the 'kick in' mic, as you otherwise might encounter a very uncomfortable amount of cymbal spill. EQ'ing the 'kick out' track in parallel to the 'kick in' like this can create comb filtering issues when using a regular minimum phase equalizer. Consider using linear-phase equalization if processing in parallel, ensuring that no unforeseen issues come up.

Compression for this track is very straightforward. Simply bus it with the 'kick in' track and use whichever dynamics setting works on the kick globally. This will help unify the envelope of the two tracks, and glue them together within the mix.

Kick transient design

In those circumstances where traditional compression fails and your kick is still lacking a little something, it might be a good idea to bust out the old SPL Transient Designer. This is a great way to emphasize the transients on the kicks without the negative effects of over-compression. Try it for yourself, and bump it to taste. I use it as a last resort, but at times it can give you *just* the punch you need.

Kick sampling

The operative question to ask when augmenting a kick with samples is... how much? How much sample do you want, and how much real kick performance are you willing to trade for it?

For many metal sub-genres the outright answer here can be 100%. At times it’s desirable to simply have an extremely consistent, robotic kick. Bear in mind that this comes with an added need to automate closely in order to keep it gelling with the mix as the tempo and intensity of the performance varies.

As far as sampling goes, the kick is the most viable element to replace entirely and still fool the listener into thinking it sounds natural. This is because the kick is mostly about the feel and punch. Using samples gives more consistent punch, and as long as you keep a reign on the high-end from getting too robotic, you can get away with murder, so to speak.

Experiment with getting the sample to follow the dynamic curve of the original performance somewhat. Drummers will tend to emphasize certain downbeats and it can be quite a musical effect, if retained in moderation. Ensure that your kick sample is in phase with the original kick (if blending). Sliding phase around can almost be like another EQ control, but the ideal spot will give you the most bottom end integrity. This is especially crucial if you blend a lot of samples into the one sound.

A clever use of kick sampling is to simply use the low-end content of a consistent sample, and combine it with the attack of the natural kick. This way we get the low-end

consistency commonly lacking in a natural performance, with a natural upper midrange attack, all the while not sounding robotic. Simply set up a steep low-pass filter on your sample track, and set it to keep low energy from whichever point you see fit, while conversely setting a high pass filter at the same point on the natural kick track. This way you can get the best of both worlds.

Kick automation

This is a big one for both rock and metal genres. When mixing most rock music, you will find a need to lower the kick substantially for any slower or cleaner verses, in order to have the headroom left to explode into choruses. For metal you will need to automate heavily in order to stop your heavily sample replaced kicks sounding overbearing during fast double-kick runs.

The principle here is quite easy. Mix the kick's standard tone during a section of the song that represents a state of 'normality' for the mix. For rock you may want to start at the chorus, which marks the point where everything is 'maxed', and downsize accordingly from there. With metal you may want to focus on parts not littered with too many intense double-kick runs, as they will inherently push you to subtract low-end and thin out the sound too much during single-hits.

For metal automation, you will want to find spots where the kicks jump out and start to sound overbearing. You have one of what are essentially 3 options, and each may be applicable depending on how the kick is jumping out at you:

1. If the high-end attack of the kick is becoming overbearing during fast runs, then simply automate a high-shelf EQ to subtract top-end every time this happens.
2. If the low-end of the kick gets out of control during these sections simply automate a low-shelf EQ to subtract bass every time it happens, still leaving the top-end clarity.
3. If the whole kick seems like it's just too in-your-face then simply automate the volume level down during these sections. Beware, as it's almost too easy to lose the kick this way.

For rock automation, your main tool will be volume. Simply move the fader around during the verses to find a spot where the kick sits well, and let it return to its punchy, normal tone during the dense, energetic parts of the song.

Some people like to use different samples for verses. Tamer sounding and more gentle, then kick into the punchy stuff on choruses. Others like to automate their compressors to ease off during lighter parts, and then put the squeeze back on during the dense parts. As always, the sky is the limit - you need to experiment as much as you can, and find an approach that works for you!

Snare

The snare is likely the most intense and difficult drum to get right. Not only does the tone originate from the drum itself, but is almost equally reliant on the space within which it's struck. The room tone is critical to good acoustic snare sounds. The snare is such a vital part of most beats and music. In many ways, it is the most expressive of the drums, frequently containing the most subtle nuances in rolls, ghosts, flams and whatever else. This makes it notoriously difficult to keep sounding natural when augmenting with samples. The key word, after the base sound has been tweaked, is

automation. But we'll get to that in due time.

Snare EQ

Most of us will be recording at least two snare tracks. The common standard is top and bottom. The top generally gives you the body of the drum, while the bottom is added underneath to emphasize crack and add some sizzle from the snare wires.

The main thing to note about raw snares – especially the close mic'ed tracks – is that they will sound like absolute crap. Even if you're recording the best snare ever forged on the fires of Mt. Doom, it's going to take a bit of work to uncover that potential. This is normal. Don't expect to bring the fader up and hear a radio-rock ready snare with a crazy attack, and a second-long room sustain behind it. Most snares will instead just be a muddy 'thwack' - the magic is something you have to help create.

You can choose to start with EQ or compression here – whichever sequence is more pleasing to you. I will begin by running through EQ first, but that by no means excludes you from dialing your compressor settings before starting to get surgical.

Top snare EQ

Start with filtering. A HPF at 70Hz is a good starting point. If you are going to get a lot of high-end from your snare bottom track, then you may want to low-pass here; anywhere down to about 12kHz. This may have the added benefit of rejecting some super high cymbal bleed.

From here, listen to the tone of the snare. If it rings more than you would like, then you've made a mistake in tracking and not used enough dampening on the snare head. In the case that you want a tight, dry sound from the snare, one of your main options is to go in with tight EQ bands and notch out the offending ring frequencies. With snares these are generally to be found anywhere from 300 to 900Hz. Essentially, the entire low-midrange spectrum.

Once you're satisfied that the snare is ringing (or not) as much as you like, you can get into the broad sculpting stage. Snares inevitably need a fair chunk of low-mid reduction. A lot of the time they will sound muddy across that very same spectrum where those rings can be found. Try to isolate areas where the snare is particularly muddy, and take them out – just enough to not kill the tone of the drum entirely. If the drum sounds 'tubby', then chances are your problem lies between 200Hz and 500Hz. If the snare sounds flat or two dimensional, then chances are the problem lies between 500Hz and 1kHz.

Now that we've largely 'de-junked' the snare, we're likely left with a more palatable, albeit somewhat weak and non-distinct sounding snare. This is the part where we start to enhance fidelity. The bulk of the low-end from a snare tends to lie between 100Hz and 200Hz, depending on its construction, the head used, and the tuning. You get two options here. Do you want to use a peak boost, or a low-shelf? Well, the answer depends on what sort of snare tone you're trying to achieve. A peak boost will generally give you a tighter, more controlled low thump, whereas a low-shelf will just lift the entire range of lows across the snare evenly. The shelf generally sounds 'fatter', but takes more space and is likely to excite some muddiness, whereas the peak boost will generally sound tighter, but smaller. A good starting point for the low-shelf is around 100Hz. Sweep it around and jog the gain until it sounds like it has enough thump for you. For the bell boost, you can basically sweep it across the 150 to 220Hz region to find something that hits home for you.

So, our snare finally has some body, and the mids no longer sound like someone choking on an oyster. One thing we have left to do is to excite the highs somewhat, and make sure our snare cuts through our (hypothetically) dense arrangement.

Similar principles to kick processing apply here. The 6kHz to 10kHz range is good for boosting the air. Just be mindful – this is where hi-hat bleed can kill you. One of the major drawbacks to processing raw snares is the amount of bleed that will start coming through. You may have to make a compromise here if you don't have as much isolation as you want from the cymbals.

The 2kHz to 4kHz range can be great for cutting through. If your snare is getting drowned, run a peak boost around here and see where it sits nicely. Things can also get quite painful and 'hard' sounding here, so don't over-do it – it's a sensitive register for the human ear!

Top snare compression

Once again, very similar principles apply as with the kick drum. Slow attack, with a moderately fast release, and a ratio ideally anywhere between 4:1 to 8:1. Attack times between 10 and 30ms will generally work well for you. Release times between 50ms and 100ms generally work equally well.

Be very careful with just how much you compress here. If your isolation isn't good you can end up with more hats in your snare top mic than actual snare. If bleed is a big problem and you need to add punch to the snare, try using a transient designer as your envelope-shaping tool of choice instead of heavy compression.

Bottom snare compression

We're starting with compression on this one, because it's so vital to the overall shape and sound of this mic. Bottom snare is a bit tricky, because the snare wires are quite prominent and get harsh easily. So what we primarily want from this mic is to add some sizzle and edge to the snare tone. We start with fast attack, fast release compression. You might want to hover around 8:1 ratio. We want to slam those wires down and increase their sustain substantially. I shoot for a significant amount of gain reduction

here until there isn't much transient left, and it's mostly a wash of sizzle to complement the main (top snare) tone.

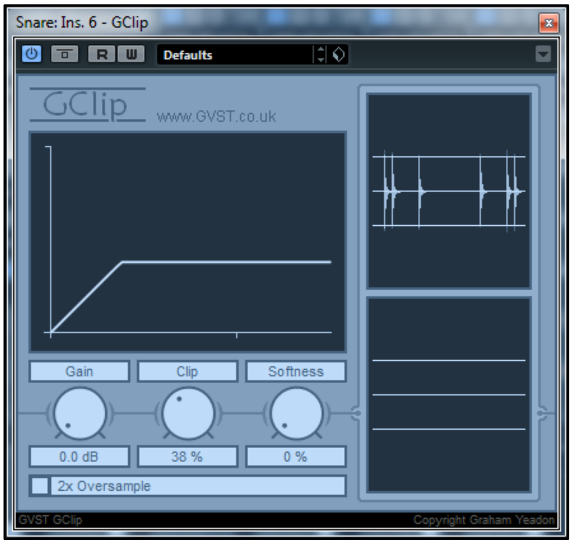
Bottom snare EQ

Start by filtering, much like the snare top track. 90Hz is a good one for me here. Feel free to dime the highs or high mids. The tone of the snares will determine what you can get away with. If you manage to make them sound airy, rather than harsh, then you can get away with blending more of them into the mix. As it stands however, think of this largely as an auxiliary mic to the top snare track. Blend in as much edge as you think it may need. A peak boost around 5kHz may get you started, but feel free to jog it around and see what works for you. You may find a shelf is less harsh and better for your style. Just watch out, as boosting too much broadband treble can make this get harsh, quickly.

Feel free to use this opportunity in order to boost more low-end into the snare. A peak boost anywhere between 150Hz to 220Hz could be beneficial and add some extra body without exacerbating the muddiness found in the top snare track.

Snare limiting

This is a fairly important one if using a dynamic source. Snare tracks are usually quite dynamic, even with more consistent players, and certain hits can get quite harsh and 'pokey'. A quick way to get them under control is to use a limiter on the back end of your snare processing chain. Set it so it only acts on the offending hits, and leaves the majority untouched. It's very easy to castrate the punch of the snare here.



Snare clipping

This is a technique that has been popularized in recent years in order to get around the effect that brick wall chains in mastering have on snare transients. As the snare is traditionally the first thing to get lost after mastering, this was discovered as a shortcut way in order to retain its transients after maximization.

The process uses hard digital clipping to chop the top and bottom of the waveform in order for it to use less headroom. Now instead of a master limiter squashing the originally large snare transient downward whenever it hits, it instead lets the chopped peak through. The D/A converter panics because it cannot recreate the chopped waveform (a pure square wave doesn't exist in the analogue realm) so it essentially overshoots and creates a faux-peak, which retains the illusion of a punchy snare on output.

Crafty? Yes! But, very easy to over-do. Only use this sparingly – ideally while monitoring

through a temporary master chain of your own. Too much clipping and you will castrate the attack of the snare.

Snare transient design

The transient designer can be a great tool for snares. It's a great way to emphasize the transient without the negative effects of compression (i.e. bringing up the cymbal bleed). You can even use the release portion to emphasize the life in the snare if it's exceptionally dead. Just beware that doing so can bring up unwanted floor noise and resonances you may dislike.

Toms

Processing toms lies somewhere in between the methods used to process kick and snare. The way they sit on the frequency spectrum makes them behave in a very 'bass-drum-like' manner. Depending on their construction, the skins and dampening used they can have quite differing tonality. The idea is to work out how each tom will sound best within your mix, and as part of the drummer's style of fill-playing. This way you will be able to leave enough of a hole for them without blasting out the few remnants of headroom in your mix every time they are hit. They can be notoriously difficult to trigger accurately and also have quite a pronounced 'fake' sound if completely replaced.

Tom EQ

Much like kicks, toms invariably have some mud in the lower-mids, and resonance in the mid-bass frequencies that can play havoc with your mix. How much and where it's all centered will depend on many factors, ranging from the size of the tom, its construction, the head, tuning, position and angle of the mic, as well as how it's hit.

A good starting point with toms is filtering, as usual. My high pass filters on toms tend to range between 50Hz to 90Hz. This is a very subjective thing and will depend on how you personally like to hear toms in the mix. If you want them shaking the floor, then you may want to consider lowering the filter down to 40Hz. In general I try to keep the higher pitched toms high-passed further up than the low-pitched ones, in order to let each resonate somewhat in its dominant zone.

Let's look at the low-midrange now. There is generally a collection of mud between 200Hz to 400Hz on toms. A wide scoop normally does wonders down here in clearing up their tone, and how nicely they play with your mix.

The next area to look at is between 600Hz and 800Hz. There is usually a flat or '2 dimensional' quality to toms stemming from here – especially if the tracking room was small. Feel free to take away some content here in order to let the toms breathe more. Sometimes you have to get fairly brutal with your subtractions in order for them to sit right. Toms can have very hefty build-ups of undesirable mid frequencies.

After this you will find that 6kHz to 8kHz is a good range to start with for high-end boosts. It's essential that you have the overheads in while doing this because they provide a lot of the air to the toms, and you may end up with something brighter than you intend by processing without perspective. If the toms still don't cut, then consider our familiar 2kHz to 4kHz range to get them through. This will boost stick attack and really help get them through that wall of guitars.

Lastly, if you find your toms are still overly resonant and not sitting how you'd like, take a look at the 100 to 200Hz range. They will frequently have resonant peaks here, which can be helpful to subdue. Using this in concert with a low-boost will help create a larger and less obtrusive tom sound. For low-end, either try the default shelf at 100Hz, or simply run a peak boost between 50 and 120Hz (depending on the size of the tom, and where it needs to punch!).

Tom compression

Nothing out of the ordinary here. Similar principles to kick and snare. Slow attack, moderate-fast release, around about a 4:1 ratio. Good compression range for toms tends to be between 3dB and 6dB. Try compressing them again on a tom-bus for extra control. If you're finding that the mid-bass or low-mid frequencies are out of control, yet you can't subdue them without killing the body of the toms, then think about trying a multi-band compressor on the tom bus. Set it so that it acts across the offending frequencies, and take out as much as you need, while bumping the make-up gain to keep the body of the toms intact.

Tom limiting

Limiting toms can help maintain control over errant transients, much as limiting snares does. My personal preference is to hit the toms on a bus with an FET compressor to do a majority of the controlling work. This has the additional benefit of adding some density to the toms. Whatever is left over gets controlled by an L1-style brick wall limiter.

Tom transient design

As with kicks and snares, you can gain some 'free' transient punch from toms by using a transient designer. A good place to use one is on the tom bus, where a single stereo instance will affect the whole lot of them uniformly. Try to save this technique only for weak hitters, and poorly captured toms. In general you should be able to get a solid tom sound without too much post processing trickery.

The room

For your average metal mix this is going to be one of the most non-essential tracks you have. It can be used to add a sense of glue and 'togetherness' to the kit, but otherwise

will only serve to add mud and junk. For a pop/rock mix, this microphone is almost essential. It contains the crux of glue for your kit, and will also be a potential source of extra low-end for your drums.

Room EQ

The very biggest problem you will run into with most room tracks is cymbal spill. They will seem like they're coming from everywhere. As such, you may want to try a substantial low-pass on them. Think of this as an auxiliary set of tracks. You are trying to create a background wash, and a sense of glue to unite the kit and give it a greater sense of density. Most of your work with EQ is to remove the mud that builds up in the room, take out the cymbal craziness, and emphasize the lows.

Start with wide cuts throughout the low-mid spectrum in order to clean the tracks up. 200Hz to 400Hz may house the majority of your muddy junk. Another point of interest can be up around 700Hz, which houses a lot of that 'flat', 'papery' sound.

The important part here is to crank the lows. Turn up a low-shelf around 100Hz and let the room mics fill out your drums' low-end. The kick will take on more of a dense, pillowy characteristic, and this will help you in filling out sparse drum parts with extra tone and sustain. Use a high-pass filter to counter-act your boost and get the low-end sitting just right. It should go without saying that if you're dealing with a dense drum arrangement with a ton of double kick parts, you do not want to overdo this, as it will completely destroy the intelligibility of your low-end. Be wise – use only as much as you need. Slower, 'rockier' songs are much more conducive to the type of frequency content room microphones tend to provide

Room compression

Great starting point here is the 1178, 4:1, 8:1 or all-buttons in mode. Attack anywhere from noon to 9 o'clock, release full clockwise. Slam the room until it's vibing just right for you. Be wary however, this is where the cymbals start to become a real problem. After you're done with your slamming, you may find you need to get more extreme with your low-pass filter, or simply adjust to the idea of using the room mics for some of your cymbal tone. You could also choose to be a bit more conservative with your room microphones and treat them as an actual part of the kit tone, rather than annihilating them with your 'all-buttons mode' until they're just a thick wash of distortion that creates density and sustain behind the close mic tracks. In this case you can simply compress the room with a more common ratio of 8:1, and stop short of turning it into an unintelligible mess.

Cymbal spot mics

The general idea with these is to complement the overheads. You can leave a bit more body in the direct tracks and bring them up under the overheads to create a sense of solidity. With spaced pair overheads these spot mics can be fantastic in defining a solid spot in the stereo image for a cymbal to sit.

Always try to process these with the overheads going, as they are solely support mics.

High-passes on these can range from 300Hz to 500Hz. After this you might find yourself removing some mid junk from 500Hz to 700Hz. Always be mindful of the critical 2kHz to 4kHz vocal range, and make sure these aren't eating into it. From here it really depends on what the overheads need. If they are excessively bright, then it may be a good idea to keep these direct tracks a bit duller. If you need a bit more sizzle, then perhaps consider pumping the highs. Bear in mind that spot mics tend to collect a lot of junk due to their close proximity. Don't feel bad if you need to cut substantial mids on these tracks, or have to do a few - even several - notches to take out high frequency resonances. Sometimes you're well off using a low-pass if simply trying to add weight to the overheads. Taking out everything above 12kHz should take care of any sizzling, which can build up and leave you with the meat of the cymbal sound.

Compression on these is entirely to taste. I tend to enjoy slower attack compression to add some transient and attack to the cymbals. This can be counteracted with a limiter on the back end to stop cymbals like rides getting excessively harsh on certain hits.

General drum mixing

Reading through the section above, you may get the impression that processing drums is a very linear, disconnected affair – but nothing could be further from the truth. Drums are extremely interactive, and changes made to one track will oft-times affect the perception of others. While mixing, I am constantly thinking about half a dozen of these interactions at any given point in time, and you need to develop your ability to do the same. You need to hear the problems, isolate where they are, and fix them, thinking in multiple dimensions – almost 'feeling' your way through the process.

Parallel compression – yes or no?

Parallel compression can be very handy in adding sustain to drums. It typically doesn't work as great on cymbals as it does on shells. Snares are notorious for their love of parallel compression. It can be used to add a great deal of sustain to their tone, and immediately adds a 'rock' character to the sound. Kicks can benefit from parallel compression too, as it rounds off and enhances the thump on the low-end.

It should be needless to say that if you're mixing very fast, technical metal music, you'll frequently be better off circumventing parallel compression entirely. It thrives on increasing the density on drums, whereas with these technical and/or fast genres of metal you're after tightness and precision. In this case the compression would be a quick way to turn your drums into a complete mess. Most other forms of music – especially ones which favor heavily 'natural' drum sounds tend to love parallel compression. It helps fill in the empty space, and generally makes the drums sound more textured and lively.

The general method for dialing in parallel compression is to send all the drums you want parallel compressed to a compressor acting at extremes. Anything from medium to fast attack, and very quick release, high ratio and a LOT of gain reduction. Make sure the drums are really being annihilated here. You then raise the level of that compressor in the mix until it just barely starts to affect the drum balance. Once you reach that point,

stop. Now mute and un-mute that compressor, until you're hearing the subtle benefit and cohesion it's providing to your drum tracks. Usually this is just enough to have in your mix.

Preparing samples

It may seem like we haven't spent much time talking about the role of samples in this guide. Up until this point, that's true. The intention was to largely focus on how to get the most from the raw drum sounds before we proceeded to augment them (if necessary/desired) with samples. One of the most exciting aspects of drum mixing is preparing custom samples for each project. Even if you have a bank of 10 samples, their potential combinations with one another open up a whole array of possibilities for you.

What I would advocate doing is preparing the samples for your project ahead of time. For instance, I commonly run a 'premix' session before I begin the mix on a record, with the entire album loaded up in there. I will use this opportunity to get a rough mix going with the drums, and then experiment with which samples may work in tandem. Once I find a great sample that forms the base of the interaction with the raw drum, I will then go on and try to keep adding to it by finding other samples that add something desirable to the tone. So it's quite common, for instance, to mix a very low 'thud' snare for mega low end, with a higher piccolo snare for cut. Both contain elements neither would be able to possess all by itself in the real world, so what we're trying to do here is create a 'franken-sample' which has the best aspects of everything for our record.

A common question is, "do you just line all the samples up and process them together, or do you filter each individually for only its desired characteristics"? My answer is: whichever you want it to be. If you are set on filtering only what you want from each drum, that's absolutely fine, and will give you a lot of precision in the sound you tailor. Personally, I tend to stack them, full-range, until I notice something about an individual sample that bothers me. Let's say I like the attack of one kick, but the overly resonant low-end isn't gelling with the rest of the drums. Well at that point, you simply high-pass until you're left with only what you want from that drum.

After all the samples are lined up and worked out, if I have the time I will endeavor to process each individually. The aim is to de-junk and focus it into its dominant zone, creating a clean, mix-friendly sample. The other option of course, is to just let them all combine, and process them on a collective snare bus. Ideally, I do a bit of both for maximum flexibility. Samples really let your imagination guide you, and create a whole new layer of fun to drum mixing. This is where things can start to sound really huge.

Just remember, if you don't like the way your sample blend is sounding, don't reach for the EQ! Either rebalance them in level, or mix up some different samples. It's always better to mix with the raws and levels, rather than processing. This way you can use EQ and compression only as the final element in getting the tone where it needs to go.

Sampled rooms

This is a topic that comes up often. Sampled rooms are a true wonder and a blessing for those of us working in the digital age. As we found before, one of the biggest limitations

to our acoustic room sounds is the cymbal bleed that creeps up and messes up our mix. With sampled rooms, we no longer have that problem. We can process them to our heart's content, and extract only the ambient qualities we need for our shells.

Another great aspect to using sampled rooms is mixing and matching. The room ambiance doesn't have to be of the same snare you have going in your mix. You can combine some really thuddy, low, direct snares, with a very high room wash. It's a mix and match exercise until you get to where you want to be. The processing principles are similar to acoustic rooms, except you no longer need the low-pass. You can compress all you want in order to get all the body and ambiance into your direct drums that you require. The dense, early reflections provided by these, combined with the thickening effects of parallel compression, and the lushness of a good reverb are the elements that start to turn a drum sound into something you can be proud of.

Drum reverb

This one is really all down to the individual. In general, drums tend to like colorful plate reverb. Plate reverb is commonly really dense and vibey – audible as part of the music itself. It can really help snares come to life, above all. Beyond this drums can take anything from room to hall reverb. If your tracking room was less than ideal, you can add a sprinkle of reverb to the overheads and room microphones. After this you can beef up the close mic tracks.

Dense mixes commonly eat reverb up, so you can have a fairly wet drum mix and it may not translate as such in a dense rock or metal arrangement. With that said, in mixes where I want a lot of ambiance, I tend to favor rooms and room samples over artificial reverb. The artificial reverb only tends to come in for the long tail that isn't possible in the real world. You can color and trigger verbs by sending different samples to them. One trick is to use a snare sample solely for the purpose of triggering the reverb a certain way, but otherwise having the sample itself muted. Beyond this you can feel free to EQ the reverb return to your heart's content. It's another mix element, like anything else, and can be shaped to fit the scope of your landscape better.

As a footnote, make sure that you always have a high pass on your drum reverbs. This will help keep your drums clean and intact. Anywhere from 150 to 200Hz should do.

Drum Saturation

One aspect of processing drums which we've not talked about to this point is saturation. 'Back in the day' it was quite common to record drums on multi-track tape machines, and in the process of recording them hit the tape quite hard. This had the audible effect of adding nonlinearities to the signal. You would effectively soften the transient of the drum and thicken its overall sound, bringing out more body and midrange. This is an effect which is highly sought after in digital processing, where drums can generally sound very 'pokey' and disconnected in their raw states. We touch on what is appropriate here in the chapter dedicated to Saturation.

Frequency spectrum summary

This is a summary of many of the frequency regions we spoke of, and how they relate to most drums. As always, no hard and fast rules, but drums do generally have common problem zones that can be tackled in a similar and uniform fashion each time.

22Hz to 70Hz:	The lowest of sub frequencies. Can be filtered out on snares and toms (though toms can go lower). Kick punch will ideally lie somewhere here. Most commonly between 50 and 70Hz.
100Hz to 200Hz:	The location of snare 'thump'. Can be boosted for more balls on snare. Can contain mud in kick and tom tracks.
200Hz to 400Hz:	The 'mud' region. This is almost universally true for all drums, from kicks all the way through to the room tracks. Scooping in this area will aid in clarity.
600Hz to 800Hz:	The 'flatness' region. Scoop this in order to add more depth to drums.
2kHz+:	Everything from here on boosts stick attack or cymbal presence.
2kHz to 4kHz:	Boost on kick, snare & toms in order to cut through dense mixes. Cut on overheads to stop conflict with vocals.
6kHz to 10kHz:	Good for adding top-end clarity to anything.
10kHz+:	Sizzle and air. Watch out, things can get somewhat harsh and weird up here.

Chapter 3: Apocalyptic tractor grind (good bass tone)

The one thing that, above all others, seems to define a truly ‘professional’ mix for most people is a tight, solid low-end. The search for this can be fairly arduous and involves not only questing for very good gear and technique, but also some great monitoring, and the know-how to combine these elements and come away with a whole that is greater than the sum of its parts.

If the drums are the framework of a mix, then the bass is undoubtedly the glue. A lot more important than most give it credit for, the bass forms the important union between the kick and the midrange elements of the mix. This is one element you’ll want a good monitoring system to mix. If you don’t have a monitoring system that would qualify as at least ‘above average’ - particularly along the crucial mid-bass spectrum, then it may be best to balance the bass on a pair of high-end studio headphones, with flat low-end response.

I personally love bass simply because it takes processing so well. The electric bass is, at its heart, a very imperfect instrument, with highly obtrusive fret clack in the presence region, a huge over-abundance of low-mid frequencies and a relatively uninspiring low-end, provided we’re just listening to flat, unprocessed DIs.

Imagine yourself as a personal trainer, freshly hired by a client who is motivated, but otherwise a complete weakling. You know the road is long, but you can see the genetic potential, and you know with enough squatting, benching, lean steak and repulsive raw

egg-shakes you can turn the dweeb into a muscle-mound. That's very much the story with bass. You start with something that inherently sounds somewhat unimpressive, but so long as you can identify the potential, and work towards unlocking it, you'll have your work cut out for you.

Tracking, editing & amping

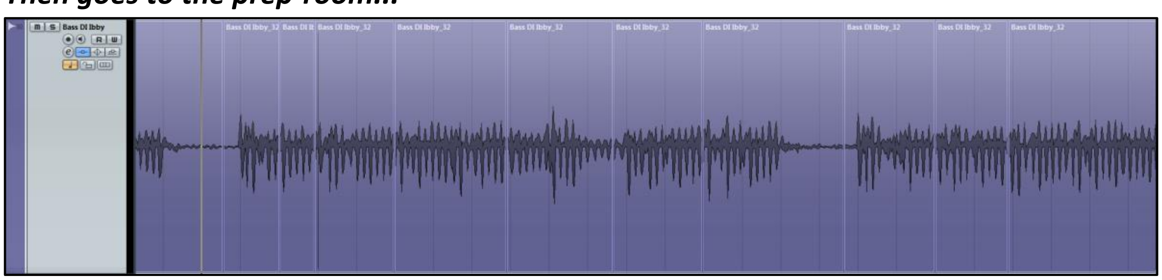
It all starts at the source...

Bass, perhaps more so than any element beside vocals, is almost completely reliant upon the source tone. Sure, raw DIs will start fairly average no matter how great the bass, but you *must* be able to hear the potential within in order to work with it. Much like trying to turn an ectomorph into a bodybuilder - it just isn't going to happen. The tides of fate just didn't intend it to be. Likewise with bass, it's crucial that the instrument be of a sufficient quality, and that the strings be brand new (cannot stress that one enough!). With a good, aggressive player the strings will start to lose that special something between 2 to 4 hours. That's your average tonal sweet spot. If you're tracking rock or metal, the bass player needs to have some serious conviction in their playing. Some might dispute this - but in my experience the sound the instrument makes in the real world is disproportionate to how it tracks. Fret clank is generally good for metal bass, as it gives you a lot of midrange to work with in order to support the guitars. Think of it as added potential. If you don't like fret noise, just low-pass the DI aggressively. If you want something to distort, well, the noise will always be there for you.

One question that comes up often involves people asking how they should set their bass when tracking. What blend of pickups are the best and what on-board EQ settings should one have on the bass? Well the answer, as with many of these questions, unfortunately isn't a sure-fire one for every situation. Ultimately it depends on what tones you are chasing, and how close to finished you want your DIs to sound coming out of the instrument. Personally, I prefer to leave the instrument EQ settings flat, as I feel studio EQs provide greater flexibility for processing. Having an equal blend of both neck and bridge pickups tends to be a great starting point for most music too.

If you feel you need to get something a little more impressive sounding on the way down, you can add a bit of compression and EQ with your studio gear. This can help incrementally build toward your final goal, provided you don't overdo things and wreck your tone before you've even begun. Now off you go, try to convince the bass player he needs to fork out $300 in strings to record the CD with you.

Then goes to the prep-room...



Assuming the band haven't already left your studio in light of your seemingly ridiculous tracking demands and don't think you're outright insane to boot, you're now off to a good start. You've tracked a good bass, with a good player, who has played it with conviction, and you've captured it all with a high fidelity, transparent DI chain. You may have even taken a Sansamp line or mic'ed the amp at the same time - many do, as it saves time reamping down the track.

The first thing you want to do is make sure the bass is tightly pocketed. A simple way to do this is to listen to the bass along with the drums and make sure they're grooving well together. If there is the occasional slip, feel free to cut at the transients and nudge the notes around, within reason. Making sure the bass is edited tightly before you begin mixing will do wonders for the speed at which you'll be able to achieve a tight low-end.

If you want to take it a step further you can actually fine-tune the bass notes using tools such as Melodyne. This is a little more involved and would take a mini-guide in its own right. Just let it be said that if you want to do this, it should be done before any timing-related editing of the track is performed. The algorithms which tuning software works on tend to have problems with cross fade points and time-stretching artifacts. Having said that, if you do go through the trouble of fine-tuning the bass notes, you will find the instrument will want to sit 'in the pocket' much easier during mixing. Preparation is half the battle, and getting bass sitting right is oft-times more than just wrestling with the ideal ratios of EQ and compression.

Your next step, provided you possess a boatload of patience (or a few spare bucks) is to volume-ride the bass in order to 'prepare' it for your compression chain. The classic way to do this is load your assistant up with an RMS meter and let him go to town drawing volume automation curves, note by note, or intelligible chunk by intelligible chunk. Since not all of us are rich, insane, nor own slaves we fortunately have market alternatives for this purpose. There are plug-ins out there, such as Waves' Bass Rider, which will do this job for you. In fact, they do it so well that they will facilitate your ability to use less compression on your final bass tone, thus leaving a lot more of the life, punch and envelope intact. You'll find that bass notes will sit on a solid bed of level underneath your mix, never getting lost due to picking variance or 'dead spots' on the instrument. This is invaluable in the quest to attain a tight, 'professional' low end.

Its purpose gets distorted...

Amping and raw tones are somewhat outside the scope of this guide - but given how holistic bass processing is, we need to touch on how to dial fundamental bass sounds.

Unless you are extremely confident in your abilities to pull sounds on the spot, you should always take a bass DI line, straight from the instrument. This is your fundamental starting point in many ways, and given the way mix engineering has developed over the last decade, will many times be the only element you're provided by an artist looking to get their record mixed.

Whether you do it in parallel while tracking the bass DI or reamp afterward doesn't really matter, but the fact remains that many people still enjoy getting the bulk of their bass tones from real amplifiers. This is in many ways similar to guitar tracking, in that the same principles apply. The main catch with bass is that, for most intents and purposes, you're only capturing the tone of that amp to later distil it into a very specific band of frequencies. Let me explain why.

It may initially seem impressive, mic'ing a massive fridge cabinet and achieving a large low-end from the get-go. However, once you begin to delve into balancing the low-end from the amp you will find that there are more factors at play than you originally bargained for. The amp, the cabinet, the speakers and the room they reside in all have their own resonant characteristics. This is as opposed to dealing with the DI line, which, even though at first glance sounds unimpressive, is subject to fewer unpredictable factors. At its very core, even with excellent bass rigs and great tracking skills, in the vast majority of live spaces, unless extremely large or heavily treated, you are going to be dealing with some serious room modes across the 40Hz to 250Hz frequency range. With a bass DI you are conversely only dealing with the resonant characteristics imparted by the materials of the instrument, its construction, dimensions, the pickups & the cable. There are other tone-related variables including the strings, picking/plucking location, DI box, mic preamp, converters etc. but they are all minor compared to the former. Ultimately, using a bass DI for the bulk of the low-end of your bass tone will leave you dealing with fewer unforeseen nonlinearities.

This is not to say that the low-end from a mic'ed bass amp and cabinet can never be useful - it all depends. Perhaps you are after a vibe-driven approach, and the resonant characteristics from that rig and room are desirable for you. Perhaps you want to minimize your post-processing, go with a vibe-driven approach, and the mic'ed starting point lands you closer to your end goal. In any case, it's worth noting that there are, once again, no rules. However, for the vast majority of people, and *especially* if assuming we are only beginning our journey with bass processing, it may behoove you to leave the DI tackling the low-end for the time being.

So, getting back to distilling our amped tone into a specific band of frequencies. We aim to do this because the core of our approach dictates that we filter our bass DI track into solely its low frequencies, and filter whatever amped lines, distortion boxes and the like into only mid and high frequencies. This way we achieve greater flexibility when processing the tone - treating it as separate low and high components. You will become aware why this is necessary once we delve into the finer points of tightening up our bass

sound to work as a vital cog in our mix. Bass is a multi-faceted instrument, and as such there are many ways to chase tones for it. For the large majority of projects I would suggest having 3 different tracks of bass, as such:

- Bass DI: This is the vanilla signal that comes from the bass guitar. Generally used for the low-end portion of the final tone.

- Slightly driven line: This can be taken from a bass amp, or from just about any drive unit, pedal or plug-in you can think of. The idea is to enhance the midrange meat of the tone, and perhaps emphasize and shape the fret noise of the instrument into something more mix-friendly.

- Distorted line: This can be taken from a boosted bass amp, a guitar amp, an array of distortion boxes, or once again just about anything else you can think of which generates euphonic distortion. Given that distortion is effectively shaped noise, it might be worthwhile working a cabinet, or cabinet simulation of some kind in there to give you a more filtered and pleasant sound from the get-go.

We will get to how these tracks interact, battle with each other, the mix and your sanity in the mixing section.

Then hits the road...

Simply because processing bass is such a multi-faceted endeavor and the instrument can be tackled so many ways, it's best if we just break our approach down into three philosophies.

The first implies that we are taking a vibe-driven approach. Our goal as such is to use quality, musical EQs, and good, thick-sounding compressors to chase our tone with broad strokes. We synergistically use the compression and EQ to chase our bass tone, and we don't focus on making the low-end entirely 'flat'. What we want here is something that sounds chunky and vibes well with the mix. This tends to be the method of choice for those who work with slower, groovier rock-esque music, and have an abundance of analogue gear whose natural saturation will help them chase those vibes.

The second philosophy is for those of us who are out to turn our pansy, woody, muddy electric bass guitar into something that resembles a synth bass. What we want here is optimum headroom utilization for our mixes, so the ideal bass tone is extremely even across the bottom frequencies, has an extremely small to non-existent dynamic range, and epitomizes the idea of something 'sitting in the pocket'. We use filtering, broad EQ, surgical EQ, staged compression, multiband compression, limiting and distortion to get us where we're going. This is the surgical approach - apt for those who need to create expansive, yet tight low-end, perhaps to work with very busy arrangements, or fast, technical music. It is an approach quite suited to those who work on modern metal music, using largely digital tools - which are what make such intensive and transparent processing possible.

The third approach is a mixture of the prior two, and is the reflection of more years of experience and self-restraint. It is all about doing more with less, affording more flexibility than the first approach, yet not carpet bombing the bass into submission as one might with the second.

Processing

Filtering bass

It's widely accepted that the most versatile approach to processing bass begins with filtering it into at least two separate components. This way you end up with a very pure low-end signal, unaffected by full-range distortion, which tends to emphasize farting in the mid-bass and low-mid spectrum. It also gives you the flexibility to process each element of the signal individually, if you feel you need it.

So we've got our Bass DI, Bass Grit and Bass Distortion tracks. We bus them together into a single 'Bass' group, and proceed from there. What you want to do is start by low-passing the Bass DI track. Around about 200Hz to 300Hz tends to work well with most filters. Experiment until you get satisfactory results with your own, as the order and sound of the filter may differ.

High pass your Bass Grit and Bass Distortion tracks anywhere from 500Hz to about 700Hz. Low-pass them at 4kHz or upwards. The distortion track in particular tends to need a lot of filtering in order to become useful as musical information for the mix. It's not uncommon for it to sound terrible until it's been filtered, EQ'd and blended.

You might think at this point: "Wait, won't that leave a massive gaping hole in the low mids?" The answer is yes, and it's precisely one of the effects that you're after. The low mids are a constant battle with bass guitar, and they always seem to creep up on you when least expected, so anything we can do to minimize their presence early on is a benefit. As always, use your discretion here and set your filters according to what sounds best to you. If you are anal retentive about keeping phase relationships intact, it would be wise to set these filters with linear-phase equalizers. They will have the effect of keeping the frequency relationships between the tracks linear, instead of dependent upon phase shift, and you will reduce any potential oddities which might rise up from such drastic filtering.

Equalizing & compressing bass

What!? The two main aspects of processing bass both in one section? Contrary to what you might think, I'm not getting lazy. The issue here is that the EQs and compressors are in many ways acting the same, as far as bass is concerned. Ultimately, your goal is to even out the low-end, carve the mids in a pleasant way and be left with a bass tone that's tight and punchy. EQs and compressors are used synergistically to this end.

Before we get to that, we start with our 3 tracks and 3 faders, which form the basis of our bass tone. Manipulate these until you have a balance that feels right to you. Then bus them together into a single bass group - this is where we will do the majority of our

processing.

It's worth noting that it can really help to process bass with the drums going. Ultimately you're trying to lock the two into each other, to form the rhythmic base of your mix. Guitars, vocals and synths, unless previously mixed, would be a distraction at this point. The name of the game here is staged processing. Every single link of your processing chain works to alleviate the load on the successive link.

Vibe approach

You can start by high passing anywhere up to 90Hz, which as you can immediately hear, tends to discard a large portion of the meat of the bass. It sounds tighter, yes, but there's virtually no low-end. We counteract this by either shelving or making a peak boost in the subs. Try anywhere between 50Hz and 70Hz. This lets you start the game off with the kind of sub-low you want from your bass tone. It works especially well if you are EQ'ing this boost into your first, and main compressor. The compressor will act on the low-end, and suck it back down, giving you a thick, albeit lopsided, sub bass tone.

Now that we've established how we want to deal with our subs, we can move on to broadly carving out our bass tone. Chances are, at this point, that you'll have a bass tone very heavy in low mids. Most raw bass sounds need a substantial cut between 250Hz and 500Hz (depending on the bass). Don't be afraid to go very broad and very deep on this cut. A/B with bypass after you've done it to ensure you haven't taken too much out, and make sure to match the output gain so that the EQ'd signal is the same level as the one that went in. When processing bass we tend to get so drastic with cuts that it's easy to forget to compensate with our output gain.

Now take a look at the high mids. Most times bass guitar will have obtrusive 'clack' sounds up here from fret buzz. Chances are that you will need to make some surgical or broad cuts between 2kHz and 4kHz in order to tame this. You can compensate for the lack of presence after this by boosting a bit on your high shelf to get some air back into the bass. Go easy here, and make sure you're not emphasizing any harshness.

If you have any EQ bands left, you may want to take a look at the 160Hz to 200Hz area, which is a common mud zone for many basses - especially those with darker bodies such as Mahogany. A cut here will normally clear things up significantly.

Finally, make sure that the region between 60Hz and 80Hz is unobstructed. This is so that your bass drum can peak comfortably, without clashing. It helps to EQ this region with the drums going.

Now let's look at compression. Provided you don't already have a compressor in the chain, acting and working with our filtering/sub boost method, now is the time to engage one. Or two, perhaps. Always think staged compression when it comes to bass. Share the load, and the final tone will be less lumpy. Consider starting with an FET type - something 1176-esque. Either run this into an Opto type or another FET type. The first 1176-esque compressor will do a significant amount of work. Since our attack range is between 1 microsecond and 1 millisecond, we can afford to open up a bit here and go for a medium to slow attack. The idea is to let enough of the attack through in order to

not stifle the bass, while also doing a significant amount of gain reduction. The release here is, ideally, very fast. Try a 4:1 or 8:1 ratio. Our second compressor is there chiefly to share the load. Using an optical compressor here, like an LA-3 or LA-2 style, will allow you to get a different envelope and color of compression to complement the FET. Using LA-style compressors is quite self-explanatory, as you simply turn the gain reduction knob until it sounds good, and is vibing well with the first compressor.

Provided we're using an FET style compressor here instead, the idea is more about shaving off the peaks. Set this one to work at extremes. Fast attack, fast release, 4:1 or 8:1 ratio. Shave off only a few dB at most, and use it as a leveler, in order to catch the errant peaks in the bass tone.

Finally, finish it all off with a brick wall limiter. Something very simple like the Waves L1 is perfect here. Simply clamp down until you have the best tradeoff between size and control. Our EQ, 2 compressors and limiter form the basis of our sculpting tools. Now that the entire chain is in place, mess around with the interactions of the different pieces. Adding old compressors from decades ago (especially ones with transformers) into your chain will alter the sound of the bass, and in some ways undo your EQ processing. Make sure to be monitoring with everything in place when you make your moves, and experiment balancing the load between the different compressors. Some bass tones may want more reduction on the FET. Some may want more on the limiter, while others more on the Opto. Play around until you achieve the tone most complementary to your mix.

Surgical approach

This is not for the faint of heart. One could say it is only for those born with an anal retentive nature, and those who innately possess an immeasurable amount of self-loathing. This is my approach to taking that muddy, peaky trash-bass and turning it into a tight, grinding instrument of the apocalypse.

This approach is reliant on much processing, so you will lean on digital plug-ins for the bulk of it. There may be an analogue EQ, compressor or two in the chain for fatness, but otherwise you will be all ITB.

A notable difference to our earlier 'vibe approach' is that we may want to start with compression this time and follow through with EQ in the middle and toward the end. The problem with EQ'ing into compressors is that their compression will in many ways counteract those EQ moves. The idea here is to stay clean, use each stage to marginally improve our tone, and build it piece by piece.

Start by high-passing to get rid of all the frequencies that aren't useful to your bass tone. Most of the time this will be anywhere between 40Hz and 60Hz. Feel free to go higher than you might normally go, as we will be doing a significant amount of EQ processing to the bass, and the subs will slowly creep back up as we make our cuts. You can do this only on the track that's handling the low-end portion of your tone, if you like. The rest should already be high passed enough for it not to matter. You might also consider starting off by hitting these individual tracks with a touch of brick wall limiting to ease the load on the compressors. The idea is to just shave some peaks to avoid pumping down the track.

Start with an FET-style compressor. Something 1176-like is ideal. Some of the digital variants can be even better than the hardware because they're cleaner sounding. 4:1 or 8:1 ratio, attack no slower than 3 milliseconds, fast release. Try aiming for 7dB to 10dB of reduction as a sweet spot here. If certain parts of the performance are hitting the compressor disproportionately hard then feel free to insert an L1-esque brick wall limiter before the compressor and shave off the hardest peaks (if you haven't already done so). This will ease the load and let your compressor operate more effectively with less pumping.

Follow up with another compressor of your choice. What you pull out here will be largely determined by what kind of tone you're shooting for, and what your current bass tone needs. Some need gentle leveling with an LA-2 style compressor, others need something harder to top off the peaks and some others might need a transparent compressor to take another several dB off. Whatever the case, identify what you need and go in for another round of compression. Balance the load between the two until you find your sweet spot.

Let's get some EQ happening now. If your CPU overhead allows, I would recommend that you use a high quality equalizer algorithm at this part of the process. They tend to keep the punch of the low-end more intact than the standard variety, even in face of the drastic moves we'll be making.

As before, you'll usually have a significant amount of mud between the 250Hz and 500Hz region. Do a wide scoop here. 2kHz to 4kHz once again might be a problem region due to the fret buzz. Treat it appropriately. Check the 160Hz to 200Hz region for regular bass mud (especially on darker bodied guitars). Make sure the 60Hz to 80Hz region is unobstructed so that your bass drum can get through without competition.

Now, let's get surgical.

Get your headphones on, as this is where we really get stuck into the low-end. It helps to do this with the drums running alongside the bass, making sure of course that your bass drum EQ is nearly perfect first, so that it doesn't eat into the bass and cause false impressions. Pull up a paragraphic EQ - ideally one with a built-in spectrum analyzer. Now you're both listening to and looking at the tone accurately - no shoddy room modes are going to pee in your soup anymore. Really listen to the bass and the notes as they're being played. Listen for inconsistencies and notes that jump out above others. If

you're finding particular notes constantly jump out of the bass track at you, and the analyzer confirms this with an errant peak, target that region and clamp down on it. Really jog the Q of the equalizer to make sure you get just what you need, and not a single frequency more. Be very diligent and careful here - you want every move to mean something, and you don't want to waste any precious frequency energy.

How is it sounding now? By the time you've finished juggling it should be approximating a bass tone as you know it on a record. If it still needs work, feel free to instantiate another paragraphic EQ and go in for some more carving. Try not to repeat yourself. If you're targeting an area that you already hit with your prior EQ, simply go back and get more drastic on that one. You want to maximize the amount of bands you have targeting different areas, so by the end you'll have a frequency slider for just about every frequency range you could want to enhance or subdue.

Chances are you'll have some flat stuff still going on between 500Hz and 900Hz. This is the cardboard zone - if your bass just sounds flat and won't gel with the midrange of the mix at all, try tackling this region. Make sure your high mids haven't snuck back up on you after all the compression. Be vigilant of the 180Hz to 250Hz region. Mud can creep up here when least expected, and contribute absolutely nothing of worth to your bass tone aside from wasted energy. If need be, go back and make your high pass more drastic in order to tackle the sub lows that may have popped up.

Now, you're at phase two of bass processing. So you might ask, 'What more is there to do?' Well, a fair bit - we're only about 50% done. If you've had too much at this point and want to quit, that's fine. It simply means you don't possess the adequate level of self-loathing to use this methodology. Simply skip back to the 'Vibe approach' and use that instead!

Here is where we put to use one of our most effective tools: the multiband compressor. You may think your bass tone is fairly tight at this point. Believe it or not, no matter how tight it presently is, the multiband compressor will make it *even tighter.* You will be using two bands for this. One to tackle the sub lows and the other to tackle the mid-bass. I generally set the crossover point anywhere between 70Hz and 100Hz, depending on what the tone needs. The higher band tends to fall off anywhere from 180Hz to 250Hz. You now have a nifty way of being able to increase or decrease two specific bass bands at whim. Set the attacks and releases quite fast on both bands. Move your thresholds until the tone gets noticeably tighter. The trade-off is that you will also make the presence and thickness of the bass smaller here. You can compensate for that with the makeup gain on each band. I normally run them anywhere from 0.5dB to 1dB to compensate for what's lost.

Finally, you want to top this off with a limiter. L1-style brick wall limiter will work great. Hit the bass with it until the meter is barely moving. If you're doing over 6dB of reduction on the L1 and the meter is still all over the place, then that means you aren't reducing enough dynamic range in earlier parts of the chain. Either instantiate another workhorse compressor, or hit the ones you already have open harder. The goal is to have something almost entirely flat and even. Two compressors, a multiband and a

limiter is just a starting point for this approach. It is the bare minimum. If you feel it needs more EQ or compression/limiting after this, don't be afraid to add it. Just make sure that you aren't working against yourself. Bass takes heavy processing much better than most instruments, so shape that thing to your heart's content. The more compression you apply, likely the more congested the tone is going to get from a frequency perspective. So don't be surprised if you find yourself going back to scoop out more low mid after each new round of compression. Sometimes bass takes a humongous deal of cutting to get where it needs to. We are talking well in excess of 10dB, if necessary. Don't be afraid to make it happen if you need to. If you're mixing rock or metal, you're likely dealing with very busy arrangements, and you're trying to minimize the footprint of the bass, while maximizing your effective use of headroom.

Once you've found the limit of what you can achieve, it's helpful to memorize what you've done, then strip your processing chain and start again, except this second time trying to get to the same spot with fewer moves. Less guesswork means more purpose, and more effective moves. Many times I've improved on bass tones I already thought were great by doing this. You simply need the patience, and as always... the self-loathing to stick with it.

Moderate approach

Wisdom often comes with time. To this point our methodology of processing bass has been split up into two distinctly different approaches. In the years subsequent to this text originally having been written, my own methodology regarding certain mix elements has changed. While in many cases either of the prior two approaches would be applicable, I found over time the desire to attempt to do more with less. The vibe approach often felt very bombastic and minimalistic – highly contingent on the raw tone, while the surgical approach often felt over-the-top.

This 'moderate' approach can be thought of as a refinement – a unification of the two previously prescribed methods, if you will. It involves an intelligent application of our tools, in order to create a consistent low-end, and strong midrange presence, all the while minimizing the severity of processing we need to commit to at any given stage.

Let us assume that we're sitting here with our three constituent, filtered bass tracks in front of us. The raw bass DI track is what you want to work on for this section. If not done already, consider using a bass leveling tool such as Waves' Bass Rider to even out the dynamics of the performance. After this, kick in a high-pass filter, which would complement the low-pass filter we would have already instantiated as per the first section of this guide. Generally a steep filter is best, such as 24dB/octave. The center frequency should vary from project to project, yet around 50Hz is generally a good starting point. You want to reject the needless sub-low frequencies which will eat up headroom in your subsequent dynamics chain. Speaking of which, you will follow this up by running a transparent compressor. There are many clean digital compressor plug-ins which would do the job just fine, but if you have access to a clean outboard compressor such as a Distressor, consider using it. Dial it in a way such that it retains as much punch and impact from the bass as possible. This means a medium-slow attack and quick

release, with a ratio no higher than 4:1. You want the gain reduction to be substantial enough to control the bass dynamics, yet not so strong as to stifle the bass, or create audible artifacts. This stage, along with the prior bass leveling will constitute the bulk of our dynamic control, so make sure to spend some time and get it right. What you are doing here is preparing the bass for your mix chain. You are staging your processing in a way that will achieve maximum impact, all the while not working against yourself down the track.

We can now return to our overall bass group. We will proceed with 3 distinct stages of processing here.

Stage one – Saturation

This is the 'flavor' portion of the mixing chain. Use whichever form of saturation sits best with you under the circumstances. For many mixes, a simple stage of console channel saturation could be fine. For others, some tape saturation may be in order. For others yet, nothing at all could work. It all depends on the level of transparency versus glue and vibe you're intending to achieve.

Stage two – Dynamics

Your bass should already be sounding quite consistent on account of the processing done to the DI track. As an extension of stage one, here you can choose whether to continue on a vibe-driven approach, or a transparent one. This is determined by the nature of compressor you choose to use. If you are after density, you can choose to instantiate an FET-style compressor such as an 1176. If, however, you are looking to keep things as consistent and clean as possible, a simple plug-in compressor can work well here. Something as basic as a Waves Renaissance Compressor is fine. As compression is multiplicative, bear in mind that you are working on top of another, previously-done stage of compression. As a result, you can be quite sparing here. This is advantageous, as you are compressing all of your tracks - including the grit - which means you will end up with enough compression to glue your tracks together, yet not so much that you stifle the midrange punch. At this stage you control how the bass will sit in the mix.

The attack control on the compressor will determine the amount of 'forward' presence, or punch the bass will retain. Conversely, the 'release' portion will determine how far into the mix the bass will sit. My recommendation would be to keep the attack between a medium to fast setting, while keeping the release to a relatively fast setting. The release can be determined based on the speed of the music, where slower material will allow you the latitude of longer release times, while on faster material you will find certain notes being stifled – perhaps too much. Try to not exceed more than 5-7dB of gain reduction. It's generally quite easy to hear when you've gone over the top here, as the bass tone will begin to sound neutered and lifeless. You want it pinned down enough to be consistent, yet still jumping around enough to sound lively. We aren't looking to square off the waveform just yet. That's for the next stage.

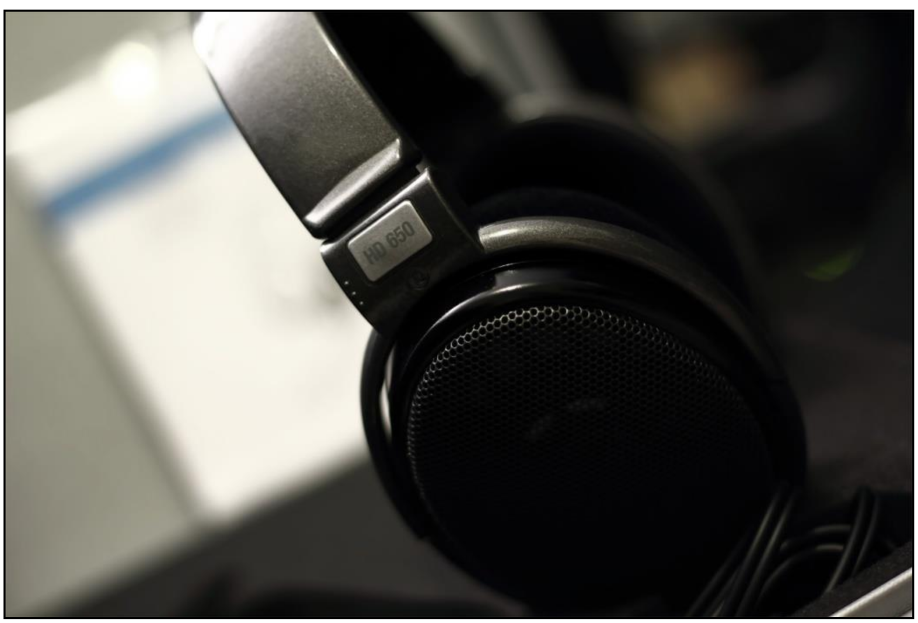
Time to bring out the brick wall limiter. An L1-type tool is perfect here. The intention is

to control whatever errant transients are left in the waveform, so that no odd peaks jump out of the mix. As stressed in prior sections, avoid the urge to go overboard here, as the L1 can destroy a track unlike anything else. Look to be shaving 1 to 2dB from the peaks every so often. You're not looking to pin the GR meter. Some passive pickups will be prone to errant spikes of excessive level, at which points you may find the plug-in metering 6dB or more in reduction, so use your own discretion when it comes to setting a reasonable level.

That's a major part of our processing out of the way. We can breathe easy now, as all we have left to focus on is our final stage. This method doesn't include the same reliance on symbiotic EQ/compression relationships as the prior two. Our approach is more straight forward now, and everything is done incrementally in order to allow us to categorize our processing into stages, thus being easily able to backtrack and problem-solve if we discover something is amiss with our resulting bass tone.

Stage three – Equalization

Begin by instantiating a flexible, clean digital equalizer plug-in. Something like Pro-Q or eQuality would be perfect. After reading the prior approaches, our common problem zones should be fairly familiar to you. The chief difference here is that you'll be focusing only on what's necessary, trying to catch as much content as possible in one move, as opposed to judiciously using as many bands as you like.



Our filtering is already complete at this point. The only thing you will need to be wary of is the sub content creeping back up as you follow the rabbit hole of EQ. If you find the sub frequencies are becoming unruly, simply go back to the high-pass filter set on your DI track, and get more aggressive with it.

The equalization phase of your processing focuses on the same frequency ranges laid out in the aforementioned 'Surgical Approach'. 60Hz to 80Hz, 160Hz to 200Hz, 250Hz to 500Hz, 2kHz to 4kHz. Between these ranges and your filters, you've got control over just

about every significant band on the bass spectrum. Be clever. If you find that the lower midrange region is overbearing on the bass, instead of making a deeper cut at 250Hz, why not simply increase the severity of the high-pass on your 'Grit' and 'Distortion' tracks, or the low-pass on your DI track? The net result is the same, except you've avoided the phase shift and deterioration of introducing another band of EQ into your chain.

Once you've made the broad cuts and begin to get more surgical, get the headphones out. Use the real-time spectrum analyzer built into your equalizer while listening for any errant notes. Instead of creating a new band of EQ every single time you hear an individual note jump out of the mix, try to catch that note with bands you already have available. If you have a cut at 70Hz to let the kick drum punch through easier, and you find there's an errant note at 89Hz, try to widen the cut you already have at 70Hz in order to catch it. If you find there are excess errant notes, and you need to use many bands of EQ to catch them all, perhaps consider that you haven't been compressing judiciously enough in the prior stages. You may need to try alternate settings, or a different compressor outright.

I would advocate occasionally monitoring your overall bass spectrum by using an averaging spectrum analysis tool. Something alike to what Voxengo CurveEQ has available. This way you can play through a chunk of the song, and see where the bulk of the bass energy is situated. Rather than the very microscopic and intense view you get with a real-time analyzer, this way you can potentially see any larger problem zones or concentrations you may not have been aware of. It's the aural equivalent of taking a step back and looking at the whole picture.

Once you've made all your moves and evened out the bass, it can at times feel a little lackluster. Don't be afraid to crank in a bit of a low shelf to bring the bass energy back up. Sometimes a shelf centered at 100Hz to 120Hz, bumping a bit of low-end can be just the ticket to a bass which is sounding emaciated after all the processing it's been given.

On first impression you may find that this approach is not all too different to the prior 'surgical approach', yet it makes substitutions and cuts in key areas to ensure that you arrive at the end product while minimizing processing at every stage. If you restrain yourself and make the most of every move at every stage of processing, you will gain so much more from your bass tone.

Effects

Bass guitar generally doesn’t take too kindly to effect processing unless it’s a *very* specific style of music. Mostly the instrument is there to hold down the meat of the low-end, and stay locked in. However, you might find that putting a gentle, widening chorus on an aux, high-passing until all the lows are gone, and then sending a bit of the bass signal there will have desirable effects for you. It can widen and increase the presence of the bass guitar within the mix, giving it a livelier feel.

Closing words

You might notice that most of the EQ suggestions regarding bass involved subtraction.

The reason for this is that bass generally doesn't take boosts well if you're searching for an even, balanced tone. You tend to be best off thinking in a subtractive manner, removing junk and excess energy, rather than emphasizing it.

Experiment with alternating the bulk of your EQ'ing before and after the compression. You'll find that EQ'ing into the compressors is a more tactile experience. You alter the way the compressors react both by how hard they're being hit, and also how the EQ is pushing into them. It's possible that the end result will be a bit messier, so if you're after a cleaner, more controlled bass tone, you can try EQ'ing after the compression. In most mixes, the bulk of a bass tone perceptibly sits between 80Hz and 140Hz. If you're finding that there isn't enough presence or density to your bass tone after all the cutting, then consider a shallow boost around 120Hz to emphasize the range of bass most conducive to mixes. If your bass tone is gelling great with the drums, but gets really muddy when the whole mix is running, then chances are you have guitars that are too strong in the low frequencies. The general rule is that the more dominant you want your bass to be, the thinner the guitars have to be and vice versa. If you want both to be dominant in the lows, then you want a muddy mix.

Frequency spectrum summary

0Hz to 60Hz:	Sub content to add weight below bass drum. The faster the music, the more should be filtered out.
60Hz to 80Hz:	The range that conflicts with the bass drum. Be careful here.
80Hz to 140Hz:	The range where bass is generally most dominant in a mix. This has to be well controlled.
140Hz to 200Hz:	Muddy, 'ringy' content is generally to be found here. Commonly needs substantial cutting.
200Hz to 500Hz:	The crux of the mud in a raw bass signal. Generally needs to be heavily attenuated.
500Hz to 900Hz:	The 'cardboard zone'. Reduce if the bass is sounding flat and 2-dimensional.
2kHz to 4kHz:	Fret noise and other harshness. Can add intelligibility to the bass in dense arrangements, but can also get abrasive if not controlled.
4kHz+:	Generally safe to filter.

Chapter 4: Poking holes in high-gain guitars

As a mixing engineer, you need to develop an intense hatred of distorted guitars. They are your natural enemy. Your arch nemesis. Heck, they're your darned Van Helsing.

No other element you run into when mixing will want to, in its raw state, eat up so much frequency real estate. Guitars chew through vital midrange frequencies like termites, they resonate over the top of bass guitars, and their very high frequencies often sound extremely harsh and unmusical.

You can think of mixing guitars as a street fight. The weaker your opponent, the more easily you can subdue them. However, the opponent always enters with a weapon and no matter how you go about it, you're still going to come off with some injuries. How substantial they potentially are will depend on three things:

- How weak the opponent enters the fight (how balanced your raws are)
- How cleanly you're able to subdue them (how well you can mix them into your arrangement)
- Whose turf you're on (how well balanced your mix is to accommodate them)

Since this is a mixing tutorial, we will already assume that you've started with a great player, who has recorded with a great instrument, which has been captured with high quality gear. If any of this isn't the case, you need to give up your search for 'perfection' now, because it won't happen. The most you can hope for is to salvage what you can using the following mix techniques, or wait for an opportunity where the aforementioned is the case.

The whole process of recording, and then subsequently mixing guitars, is somewhat contradictory and ironic, as we'll find out later. Your primary goal when tracking guitars – assuming you care, and are using real tube amps with a real mic or two - is to get a large sound. What you want for most projects is something with a fine, even midrange balance. Anything out of the ordinary, such as excessive tight peaks (fizz or midrange resonance nodes) will immediately play havoc with the clarity of the mix. So you're looking for a finely appropriated midrange as your #1 requirement. The midrange is where guitars live on records, after all.

The second element is a balanced low-end. Guitars are naturally adversarial here, as they will have cab resonances that commonly lie in areas that excite nasty room modes in smaller spaces. One of the reasons that oversized 4x12 cabs are so popular for distorted guitar sounds is that they shift these resonances lower down the spectrum, where they are perceived as more musical and 'punchy', rather than outright muddy. If you're recording in a small room, then you need to rethink your circumstances. You may well be able to get a good midrange if you isolate your microphone and cab enough from the surrounding environment, but your low-end will never be clear. No matter the amount of treatment you put into that space, there will always be nodes in the low-end that get excited and cause the vital bass/guitar crossover region of your mix to be an absolute nightmare.

Here is where we get into the contradictory stuff.

So, you've loaded your raw guitars into your mix. What alone sounded quite large and nice is now completely clouding your bass guitar, vocals, and generally making the mix sound like a mess, right? Perfect! This is exactly where we want to be. The whole idea behind tracking a 'large' guitar sound was that so you could downsize it in mixing. This is where the poking of holes begins.

Try to stay in a subtractive mindset with guitars. For the last couple of years I've rarely ever increased the gain on a single frequency band on guitars at all. Picture the ideal tone that's hiding within that mess, obscured by all that junk. Ultimately all you can do in mixing with guitars is to reveal that inherent character in its pristine form. You do this by subtracting the junk. If the raw sound is bad, then your final sound will also be bad. Guitars are notoriously poor at taking processing. Second perhaps only to vocals in this area, they absolutely need to have a good character when raw, otherwise this entire exercise is pointless.

The main issue you will find with guitars is how much parking space they demand. They're like that fat dude with the hummer who takes two parking spots, and gets away with it simply because he's the CEO's cousin. Likewise, guitars are vital to driving the energy in a rock or metal mix, and take up a whole heap of real estate. They will likely be conflicting with many major, vital elements of your mix. Your process begins by letting these other elements through.

Start with filtering

High and low passes are commonly the first stage in the sequence of processing done to guitars. This is because the filters reject the most useless of frequencies. Everything that is entirely unmusical and messy can be rejected significantly with these.

The points where you set these filters, and their orders, are entirely dependent upon the mix you're creating. A common starting point is a 3rd order (18 dB/oct) high-pass filter set between 60Hz and 140Hz, and 1st or 2nd order low-pass filter (6 or 12dB/oct) set at 6kHz to 10kHz. Bear in mind that you can be significantly more or less aggressive than this. It's simply a starting point from where you can jog the filters around until the guitars start to sound the most correct in context with the mix. Sometimes you just need a tone that fits within a very dense arrangement and you need to reduce its bandwidth appropriately. Don't be afraid of doing this if necessary. Get extreme if you need to.

Move onto basic EQ

Think of this as the sculpting portion of the process. The idea is to do whatever you need to, in order for the guitars to take on a character that's more pleasing toward the mix balance. It's immensely difficult to talk about this in general terms, as each guitar sound is intrinsically unique.

Always remember, guitars take processing very poorly. So get to where you need to go in as few moves as humanly possible. Always have that in the back of your mind, reigning you in. Do more with less. Referencing guitar sounds can be a dangerous affair. You might well be able to take on the same overall shape and balance of that tone

you're referencing to, but in the process you will lose what's unique and powerful about your own tone. It's always a better idea to uncover the potential within your sound on its own terms, before referencing. Reference only for overall balance, if possible - not for the finest details. If you are having trouble hearing the rest of your mix in specific areas, then referencing may uncover where to look, but beyond that you should still treat your sound as its own entity.

Common problem zones to watch

Guitars are essentially shaped broadband noise. They're a good starting point for frequency training, because they contain so many sustained frequencies in such abundance. It's for this specific reason that you will find yourself EQ'ing them all over the spectrum, and consequently also find them causing problems for your mix all over the spectrum.

Take a look at the low-end first. Chances are there will be a peak, or a few of them, anywhere from 100Hz all the way through to 180Hz. Smaller rooms may even induce ringing and resonance further up, between 180Hz and 250Hz. Wherever you find these peaks, suck them out! They are exceptionally detrimental to the clarity and tone of your bass guitar. Having some minor 'thump' to the guitars between 70Hz and 90Hz can be beneficial in creating movement and size, but tread carefully. Always remember that the bass guitar has a role - that is to create tight, expansive low-end, which in turn creates the illusion of enlarging the guitar sound.

After this, consider looking at the low-mid frequencies. Many smaller rooms or mid-focused amps will leave some junk down here. Usually a wide scoop of sorts can be used in order to clean them up. We're primarily talking about 200Hz to 500Hz here.

The next is one of my favorites. I call it the 'cardboard zone'. It essentially ranges from 600Hz to 800Hz (but on some amps can even be centered as low as 500Hz). This is the zone that contributes to what you hear as a 'flat' character on guitars. Generally, subduing this range on most raw guitar sounds will lead to an immediate increase in air and dimension for the entire mix.

The next zone is absolutely vital. It commonly ranges between 2kHz and 4kHz, and many times is centered close to 3kHz. This is the vital vocal presence range. This is an area that you need to balance with utmost precision if you plan to have both guitars and vocals intelligible at the same time. There are no specifics to be given here, as it really depends on the excursion of the amp. There will generally be something undesirable going on in this region with a raw amp tone, and it's your job to get it under control so that vocals - the centerpiece of your mix - can poke through unobstructed.

Beyond this region, you will mostly find isolated fizz nodes. This is extremely amp/cab/speaker-dependent and no specifics can be given. Feel free to sweep a tight EQ boost beyond this region. Listen to hear if anything stands out to you as particularly unpleasant or detrimental to the mix. You can sometimes end up with a few to several tight EQ cuts in the highs to tame fizz. Each cut will bring increased clarity to your mix, but decreased clarity to the guitars. This is the tight rope that you walk with guitars.

Consider multiband compression

Taking us back to the lows and low-mids, we find many times that sucking out too much of the resonance down here makes the guitars very thin and powerless. This is largely because there are so many non-linear peaks within this region, that we'd need multiple EQs to get them all controlled in ideal proportion - after which there'd be no tone left at all from all the phase shift craziness.

So what certain crafty engineers have pioneered is the use of multi-band compression on the mid-bass (and some low-mid) frequencies of the guitar. This allows you to tame only resonances that have crossed a certain threshold, and keep the rest of the low-end intact. This is a very good solution for cab and room resonances that tend to get excited by specific notes. It keeps the rest of the low-end relatively untouched, thus allowing the guitars to stay meaty while still being controlled.

I personally run the multiband compressor between 70Hz and 250Hz much of the time - possibly higher - provided the tracking room was small and the resonances extend further. Consider a fast to medium attack, and quick release. The attack slider will effectively control how much low-end presence you allow the guitar to have within your mix. The more you let through, the bigger the presence of the guitars, but the more you stomp on the bass. A good rule of thumb is to let through only as much as you need for the guitars to function within the mix. Allow the bass guitar to do its job, and hold down the crux of the low-end.

Consider broadband compression

Broadband compression on distorted guitars is an interesting thing. They tend to naturally be so compressed because of their high distortion but at the same time still have peaks, which are caused by the exciting of resonances within the guitar, cab, room and so forth.

Broadband compressors on guitars mostly act on low-end spikes. That is to say, they will conversely make your guitars more fizzy and edgy. They will feel more upfront and controlled, but also gain a harsh, present character.

I would advise that unless you are very experienced in mixing, avoid broadband compression on high gain guitars, as it can cause more problems for you than it solves.

Small amounts of limiting can help in the avoidance of clipping at very high master RMS levels. The limiter will act very aggressively on errant micro-peaks on the guitars, and allow you to get a little bit of extra headroom. This, once again, comes at the expense of punch and an increase in presence.

Push them into the mix

A good way of working out which guitar frequencies are clashing with the rest of your mix is to turn the guitars up until they are just slightly 'too loud'. You will immediately notice the overlapping problems. Your 'panic' sense will set in (you know, that same one you have all the way throughout mixing a live gig) and you will start correcting the response immediately. This way, you're creating holes in the guitar sound, which leave only the relevant and useful frequency information for your mix. Obviously this can be overdone, so be mindful. However, it *is* a nifty technique to fall back on when you're struggling to build your jigsaw puzzle correctly.

Make them *wide*

There is always discussion going on about how to get guitars 'wider'. It seems a phenomenon that most engineers are enthralled by. The solution is quite simple, and comes in 4 points:

1. Good EQ balancing will make guitars seem wider, due to less stuffy midrange frequencies polluting the mix. A bright guitar sound will generally feel like it's pushing outward, past the speakers.
2. Un-linked compression. If you're going to compress your rhythm guitars, avoid doing it with a stereo-linked compressor. The linking will immediately appear to narrow them.
3. Use different tones on either side. This can be done by using different amps, guitars or even EQ balancing. Be careful here, as vastly different tones between speakers can make some listeners feel woozy.
4. Use psycho-acoustic stereo widener tools. This is a no brainer, as these tools are designed specifically to appear to take tones 'beyond' the speakers. Be very careful, as these are the most prone to ruin your mix. The more you abuse them, the less mono-compatibility you will have due to phase cancellations.

Lead guitars

These work on a similar principle as high-gain rhythm guitars, except that they have the additional hurdle of needing to be heard OVER the top of the rhythm guitars, whilst still largely competing for the same midrange space. The idea is to only start mixing these once you feel the rhythm guitars are in a solid place. With the rhythm guitars taking up too much midrange real estate, you will be forced to do too much wacky EQ pushing to get the leads heard, and that won't help anybody.

Generally you want these to have a little bit more mid-range exertion than rhythm guitars. Just enough to cut through – but ideally not enough to eat into your vocals. High-passing can also be a lot more drastic here, with you being able to quite comfortably go up to 160Hz if need be. The bass and rhythm guitar are already holding down the low-end, so there's usually no need to add more clutter to it.

Broadband compression can be used to effect on lead guitars, in order to shrink their footprint in the mix, and lock them in. 'Gentle' sounding compressors like the LA series tend to work quite well. Tube or optical designs in general have potential here. The idea is to smooth the tones out, and emphasize the mids.

Consider ducking the level of the rhythms whenever the leads kick in. It's nearly impossible to Tetris them around each other entirely using only EQ, so feel free to automate in order to get the rest of the way.

If you're struggling with getting your lead guitars through the rhythm guitars, odds are that the rhythms are taking up too much frequency real estate. If you're struggling with vocals along with leads, then odds are the rhythms are taking up too much presence room, between 2kHz & 4kHz. If the guitars are sounding a bit 'telephonic', then odds are the rhythms are taking up too much space between 1kHz & 2kHz. If things are sounding somehow flat, 2-dimensional and 'cardboardy', then odds are the rhythms are taking up too much space between 500Hz & 1kHz. Work out where your problems are. Mixing to a point where rhythms, leads & vocals can fire at the same time, while being audible in perfect synchronicity is what separates the men from the boys.

Saturation

Due to their proclivity to containing abrasive high frequencies, distorted guitars are great candidates for saturation. Fortunately it's not rocket science, as simple tape or console saturation generally do the trick. Try a 15 IPS tape sim, or a vintage console emulation. Don't be afraid to drive them hard with the guitars. The natural saturation/compression effects will make the guitars both smoother, fatter and more controlled sounding in a more effortless manner than straight compression and EQ processing will.

Effects

While most of the time you'll have your rhythm guitars running dry, you may notice that lead guitars are particularly warm to being 'effected'.

Modulation

Most lead guitars like at least a little bit of chorus. It can help thicken up the tone and get them sounding naturally wider. Flanger and Phaser can be used to effect, but not often as 'go-to' tools.

Delay

Delay is almost a necessity with lead guitars. The bounce-back helps create space for them, and increases the perceived sustain of the guitar. Experiment with mono delays, stereo delays and ping-pong delays as needed for the part. You'll find each has its place. Consider some 'effected' delays too. Sometimes thinning the delay signal out or taking out its top-end allows the guitar to sit much more neatly in the mix. The idea is to stop the delayed signal clashing with the dry, dominant guitar.

Reverb

While reverb is falling somewhat out of fashion in modern production, and we would likely use delay to create our 'reverberant' effect, we can still throw the odd room, hall or chamber reverb on our lead guitars to make them feel larger. Use only as much as you need. Consider using the pre-delay to separate the reverb from the dry signal and make it all sound larger.

To summarize

You may notice that our 'common problem zones' effectively covered the entire frequency spectrum. This is why distorted guitars are our natural enemy. To re-iterate:

they are essentially shaped, broadband noise, which we are trying to get working in concert with other vital, midrange-centric elements in our mixes. It is notoriously difficult to create a mix that breathes, and has a guitar tone that is perceptibly ‘strong’ as well. The one counteracts the other, and the two goals lie at odds all the way throughout the mix. If you master the art of powerful guitar sounds that don’t intrude upon your mix in significant ways, then you’ve got a huge leg up on the vast majority of the engineering population out there. Unfortunately for the mix engineer, this is a skill that lies chiefly in the tracking stage. There is not much that can be done to overcome poor guitar tracking in the mixing stage. We only strive to augment what’s already there.

Frequency spectrum summary

Bear in mind that those are general guidelines, and by no means biblical. They detail what will most likely be the case, but as always there are no hard and fast rules. Do whatever you feel you need to in order to get your mix to where it needs to go.

22Hz to 70Hz:	Can generally be filtered without issue.
70Hz to 90Hz:	Can have some useful cab ‘thump’, but otherwise to be treated as dangerous.
90Hz to 180Hz:	Cab and room resonance area. Clamp down heavily here in order to aid the low-end clarity of your mix. Clamping down too much will leave the guitars sounding weak. Consider multiband compression.
180Hz to 300Hz:	Can contain smaller room resonances, and mud from the cab. Go easy controlling this, and only take out as much as is necessary, as it’s very easy to lose power here. Consider extending the multiband compression to this region if necessary.
Low-mids:	This entire region can contribute to mud, depending on the playing style, guitar, amp, cab and room. Treat appropriately.
500Hz to 800Hz:	The ‘cardboard zone’. Decrease this to improve the dimensionality and depth in your guitars and mixes.
2kHz to 4kHz:	Vocal presence region. Tread carefully here as it determines both the intelligibility of vocals and guitars. Finding the right balance here is a masterful technique, which can take years to develop.
4kHz+:	Generally isolated fizz spikes. Treat them with tight EQ cuts in general. As always, be careful, as sucking out too much will kill the edge and intelligibility of the guitars.
10kHz+:	Can generally be filtered out for most guitar sounds. If you want some more air, you can extend the LPF to 12kHz.

Chapter 5: No, these don't go to eleven (clean guitar mixing)

Think of this as dealing with the mixing of the acoustic guitar's retarded cousin. You've taken an instrument which is perfectly fine - sounding nice, bright and pleasant, then taken the body out of it and run it through some silly electronics. Now you've ended up somewhere between acoustic and driven guitars. You don't have the same mess of harsh noise to deal with that comes with high-gain guitars, but you're also not dealing with something as open and organic as an acoustic guitar.

A touch on tracking

As usual, the source matters a great deal. The player's consistency, as well as the tone of the instrument become quite apparent with the lessened distortion level. Even the quality of instrument cable can have an appreciable benefit to audible resonance and top-end clarity. You may find yourself wanting to trade in the full-bodied tone of hum-bucking pickups in favor of more traditional, thin and jangly sounding single-coils. This brings with it the caveat of a lot of hum and electromagnetic interference susceptibility. You must be mindful of tracking the instrument away from any source of electromagnetic interference, such as your computer, UPS, rack or cheap dimmer lights. There are few things worse than having an ethereal, soothing clean electric section in a song, only to hear the notes fade into a wash of unmusical hum and noise.

By nature, anything going through the guitar's pickups is going to be more band-limited

and mid-heavy than what you hear in the room. As a result, you'll often find yourself needing to clamp down on resonances, unnecessary midrange junk and enhancing the air of the tone. Much of this can be done with the correct amp and microphone choices during tracking, but you'll usually find yourself needing to offer a helping hand during the mix process.

Digital over tube, seriously?

Due to the very dynamic nature of undistorted or lightly distorted guitars, you'll find some seriously unpleasant peaks hidden within most tones. These are more or less an unavoidable side effect of amplifying guitars in such a way. Ironically, this ends up being one of the few situations where current amp simulation technology can yield better, and more mix-ready tones than real amplifiers. Due to eliminating much of the harshness present with real tube amps and cabinets, those sims are able to present a cleaner and more balanced tone, without so many of the harsh edges. As the need for distortion isn't so great, the various modeler drawbacks aren't as evident, and their advantages become more apparent. One of the main advantages in this sense is their lessened noise. The fact that they often do sound cleaner and more sterile than their tube amp counterparts can work to our advantage.

The other upshot from this method is that the guitar tones can often be tweaked on-the-fly - essentially letting you fit the mix with your source tones, rather than approximating with an amp, microphone and a rough idea of what your final mix will be. This can be invaluable if we're looking to lessen the severity of our mix processing.

Warm them up

One thing which clean and driven electrics really love is saturation. Saturation can come in many forms, but some of the best types available to us are console, transformer and tape saturation. Choose your desired type and drive it hard. This is important as it will help control the varying dynamics in the performance without the audible artifacts of heavy compression and limiting.

Clean them up

As with most guitar tracks, what we're looking to do during the first part of the process is to filter out what we don't need. In busy mixes you're generally safe to filter anywhere between 80Hz to 130Hz or so. The bass guitar often takes care of everything below that.

Low-passing can be discretionary, yet you might find after you've done all your necessary cutting and high boosting to make the guitars come to life, they might sound a touch brittle. This is where a gentle 6dB/octave low pass might help bring the warmth back into the guitar sound. The center frequency for this filter is entirely down to the sound you are chasing, so just go with whatever sounds best under the circumstances. A good indication is whether the top-end of the clean guitars is married to the cymbals. If they sound too dull in relation to your overheads, consider lessening the filter. If they're too bright, try clamping down a little harder.

Poke some holes in them

This is where we get to the brunt of the tone-sculpting work. Bear in mind that at this stage the guitars should already sound quite nice when playing solo. That is to say, fundamentally the tone has already been shaped by the choice of player, guitar, amp, cabinet, microphones etc. Yet odds are that once you engage the mix over the top of them, everything starts to fall apart. That's alright, because it's what we're going in to fix now.

Try to think of this process as 'downsizing' the guitars. Many of the same principles from mixing high-gain and acoustic guitars also apply here. They are fundamentally the same instrument, after all.

Start by looking at resonances between the 150Hz to 250Hz region. These are the ones prone to play havoc with the bass guitar, and undo your hours of painstaking low-end tightening work. Odds are you will find more to do here if you're using a real amplifier and cabinet, mic'ed up in a real room. There are many factors which can exacerbate low-end energy in this case, not least of which are natural cabinet resonance, room resonance and the proximity of the microphones. Remove just enough energy here to clean up the mix, but not to remove all the power from the guitars. We will control the rest in a subsequent processing stage, later.

The infamous 'cardboard zone' is a no-brainer to move onto next. $5 dollars says you will find an appreciable difference in dimension and depth with a wide cut somewhere between 600Hz to 800Hz. The key here is to remove just enough so that the guitars begin to sound three dimensional, yet not so much that they sound distant or harsh. Use other parts of your mix as a reference to get the severity of the cut just right.

The presence range is critical for clean electric guitars. Often you will find that much of the character of the tone resides here, yet if you leave it as present as the tone inherently would like to be, all your vocal clarity will disappear. This makes sense, as the presence range is where the human voice is dominant and naturally wants to sit. Jog a cut between 2kHz to 4kHz. Odds are the sweet spot will be in the middle somewhere. The moment you cut this range back, the guitars will start to sink back into the mix in a very pleasing way. Your ears will breathe a sigh of relief, as the highly sensitive presence range becomes decongested. One thing you may notice is that as you increase the cut, you also reduce the power and push of the tone. You have to balance this very carefully against your mix to see just how much energy you can get away with here.

Top-end boosting. Here is something magical which we couldn't execute quite so easily on high-gain guitars. Due to the greatly lessened distortion of clean electric guitars, much of the high frequencies are still quite pleasant and airy. As a consequence we can use this as an opportunity to pull out our best quality equalizers and dial a hefty top-end boost into the guitars. Generally anywhere between 8kHz to 10kHz as a center frequency tends to work well. Generally the more gentle and musical your equalizer, the easier you'll find it to boost higher amounts of top-end.

If your guitars still aren't sounding quite right at this point, you may have to start looking back in more depth. The moves we've already made should have the tone at least 80% of the way there. However, if it's still not sinking into the mix well consider looking at these spots:

- If you find the tone to be stuffy and congested, look at around 300Hz. Very close microphone proximity or hum-bucking pickups can exacerbate this range, which will cloud your entire mix.
- If you're finding the guitars to sound somewhat telephonic and 'forward', then consider looking between 1kHz to 2kHz. Nodes in this range can really hamper the dimensionality of your mix, and you'll know as soon as you make your first cut whether the problem was here.
- Assuming the tone is still somewhat harsh and unpleasant, you may have some errant spikes somewhere between 5kHz to 10kHz. Scout out this region and take out what you need in order to bring the guitars back in check.

As always, the name of the game is to 'do more with less'. The less moves you need to make, the better. If you find yourself making excessive cuts all over the spectrum, consider going back to the drawing board and capturing a friendlier tone from the source.

Squish 'em down

Contrary to high-gain guitars, where we seldom want to use broadband compression, on clean electrics it's almost a foregone conclusion that we'll need to. Being less distorted, we don't get the innate harmonic compression benefits that high gain guitars possess. We'll be using a compressor here for both dynamic control and tone shaping. Optical compressors tend to work exceptionally well due to their soft envelopes and gentle character. The classic LA-2 and LA-3 compressors are a great starting point for those who don't like to tweak much.

We're not looking for crazy amounts of compression. Just enough to keep the guitars 'pinned' into their place. Try not to exceed 7-10dB of gain reduction.

Revisit those lows

Just in case you aren't entirely happy with how the cleans are gelling with your bass guitar, get a multi-band compressor up. Target the range between 100hz to 250Hz and set the compressor so it works primarily on the peaks. This should even out the bottom end and give your bass guitar a fighting chance, while still leaving the punch and body of the cleans relatively intact.

Trim the peaks

To finish it all off we'll want to use a primitive, L1-style limiter to chop any errant peaks from the tone. You will generally find that after a hefty dose of LA-3 style compression, the slow attack character will have allowed the transients to become quite peaky. The L1 is your final bastion of dynamic control, where you can trim those peaks back to your liking and create a mix-ready clean guitar sound. As always, try not to go overboard as you can masticate the punch of the guitars. Set the limiter so you're just taking care of the abrasive peaks, and no more.

Effects

Modulation

Chorus is a clear winner for clean electric guitars. Popularized in the 80s, it's hard to go wrong with even in the modern era. Chorus is often best printed at the source, as running it through the amp and speaker chain can add a gluing effect, which imbues it as part of the tone of the sound, rather than as an added effect. That being said, you can also choose to use a stereo widening chorus of sorts in the mix to enlarge the presence and footprint of the guitars.

Delay

This can be used sparingly. Often you don't want the delay to be audible as a such, but more so as a depth or reverberant effect to add dimension and space within the mix. Modulated delays can work really well, as can effected ones with a telephonic or washed out character, as they don't tend to stand out as much within the mix.

Reverb

Use the full breadth of your imagination here. Just about any reverb will work, so long as it suits the aesthetic of the song and mix. Anything from room reverb to chamber to hall. It all comes down to individual preference, and how large you'd like the impression of the guitars to be. It can often help to send them to a small room reverb for the early reflections, and then a larger chamber or hall type to get the lush reverb tail and depth.

Frequency spectrum summary

0Hz to 70Hz:	Rumble which is safe to filter.
100Hz to 250Hz:	The bulk of the low-end. Prone to clashing with the bass guitar. Treat as appropriate.
Lower-Midrange:	Prone to containing some excess junk from real amp, cabinet & microphone chains. Treat as necessary, but try not to go overboard as it's easy to kill the power of the tone in this region.
600Hz to 800Hz:	The infamous 'cardboard zone'. Reduce in order to increase depth and dimensionality of the guitars.
1kHz to 2kHz:	Telephonic midrange frequencies. Reduce in order to sink the guitars further back into the mix.
2kHz to 4kHz:	'Presence' range. Contains much of the power and character of the guitars. Very prone to clashing with vocals. Be careful when balancing it against your mix, and only remove as much as is necessary.
8kHz+:	Airy top-end. Boost in order to add shimmer and life to the guitar sound.

Chapter 6: The vocal center of your mix

If we're working with just about any modern genre, whether it be pop, rock, metal, dance or whatever-else-have-you, chances are we are now at the chapter that talks about the single most important element of the entire mix.

In popular music, the element that the average listener most relates to is the vocal. Even if the rest of your mix sounds like it was blown through the backside of a rhino, you have to at least ensure that the vocal is intelligible and audible enough to not feel subdued or buried. If the listener has to strain to hear it, they will not be happy with you, whether they realize it or not. This is of chief importance, especially if we're working on music destined for mainstream radio airplay.

You can't replace the instrument

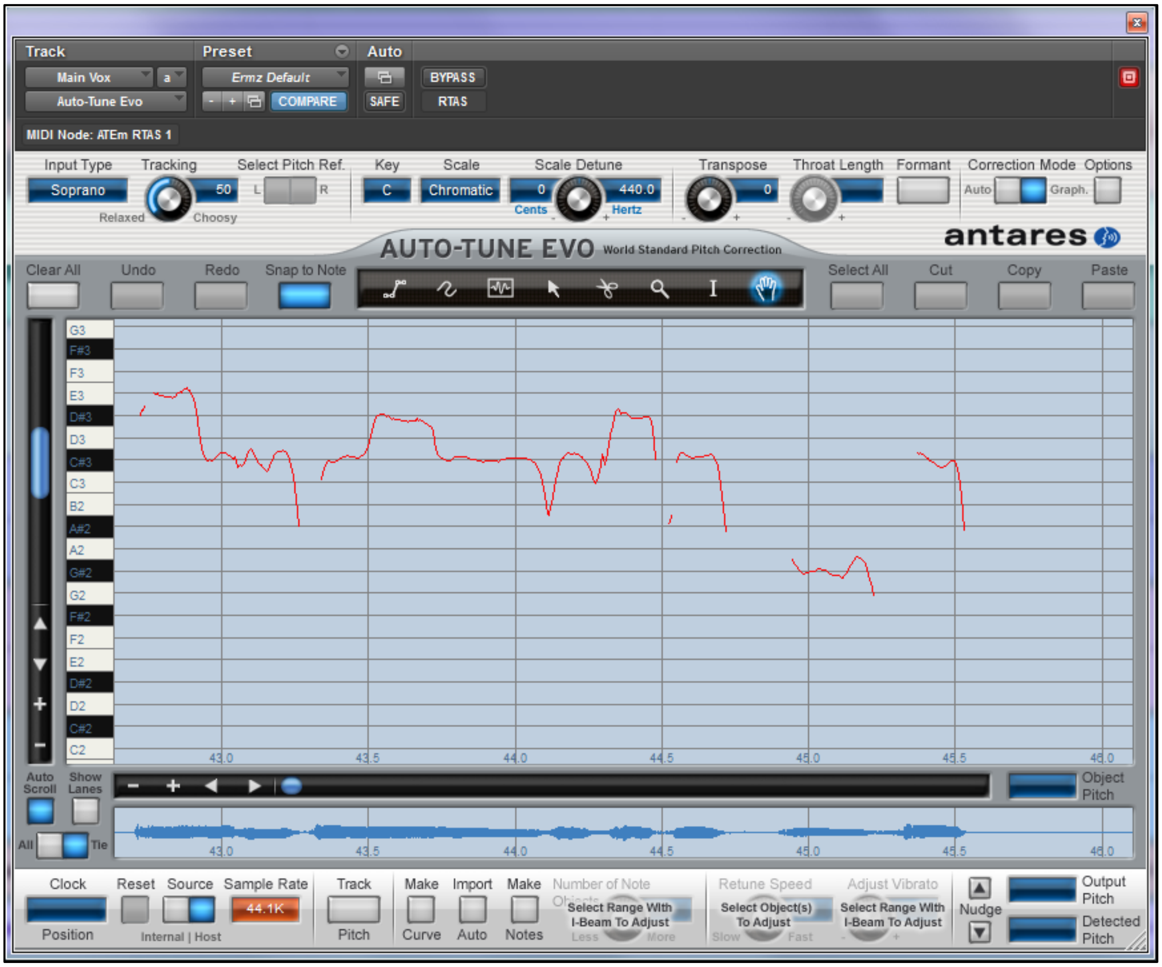
It should go without saying that unless you've tracked a competent vocalist, and done so well, you have absolutely no hope of pulling anything out here. We've all heard those arbitrarily famous rich folk decide that breaking into a worldwide singing career would be a great way to kill a weekend. What we've also heard is how hideous they inevitably sound, despite the engineers' best attempt at exhausting the world's supply of polish on it. There is no Botox for vocals – they're either tight or they aren't.

Tracking vocals is all about coaching - focusing on phrasing, enunciation, power, clarity, layering and making sure the singer feels comfortable with what they're doing. What you track is about 80% of the battle. The mix is just about pulling out and emphasizing

the best aspects of it. You cannot add anything which isn't already there. So it's either going to sound great before you've touched it, or it never will.

You can tinker with it, though

This guide deals chiefly with vocal mixing, not tracking or editing, which are two quite substantial parts of the equation unto themselves. It's worth touching on rudimentary editing techniques however, as they may get you out of trouble as a mix engineer who is looking to multi-task.



Vocal editing is a fairly big industry unto itself these days. Editing on the level that current mainstream, world-class record productions do it is an immensely time consuming feat. What you want to ask yourself here is: 'how much do I intend to edit the vocals?' If your answer is 'none at all', then that's perfectly reasonable, provided your project calls for it. If the answer is 'some', then you may just want to nudge the odd word, or tune the odd phrase, if it's called for. If the answer is 'I want it to get the full works', then read on. Generally the two things you will get to play around with in editing are timing and pitch. You will want to take care of the pitch part before you handle timing, and that's mainly because tuning algorithms tend to have issues with audio that's been spliced or TCE'd (time compressed/expanded).

Tuning begins by picking your tool of choice. The industry standard at the moment is still Auto-Tune, working manually, line-by-line, in graphic mode. This provides the most control, and also attests to why editing is a significant process unto itself. Once you've pitched an entire record that consists of over a dozen vocal tracks, you'll suddenly begin to understand why outsourcing editing looks like such a good idea to a producer.

Once the vocals have been pitched to a satisfactory standard, and consolidated, we can move onto timing them.

There are a number of ways with which to do this. The best is usually a combination of them. You can slip edit, use manual TCE, or use a unified TCE subset of your DAW, such as Pro Tools' 'Elastic Audio', Cubase's 'Audio Warp' or Logic's 'Flex Time'.

If you're going to use TCE for the bulk of your editing, make sure to use the highest quality algorithm your DAW has available, even if it has to be rendered off-line at the end. The difference in tonal quality will be very well worth it, considering that we are dealing with the focal point of a mix.

If you are slipping, make sure that you cut at the most transparent points possible. While the vocal will generally sound more natural edited this way, the transitions become the potential problem zone.

Some people will tune each word; others will only tune the ones that are audibly out. Some people will time the take syllable by syllable, and others will only nudge words that are notably out with the song. Use your own judgment, do what feels right for your project, and make the most of it.

Saturation

Being mid-range centric, vocals tend to love saturation. Anything that will take the edge off the mic and the proximity of the singer is generally a welcome relief. Tape and console transformer saturation tends to work great here. The transformers from old compressors are also welcome. What you're trying to do here, at the outset, is ensure that you have saturating elements that will funnel the vocal further into the midrange while also taking off the edge. This way, when you boost the high-end down the track, you'll mainly be boosting gentle air.

If you're using a lot of analogue gear in your chain, or mixing through an old console, chances are you're getting all your saturation already. For those a bit more limited, or digital in their workflow, this may be a good time to go hunting for quality saturation plug-ins. Start with tape and console saturation, and work your way on from there.

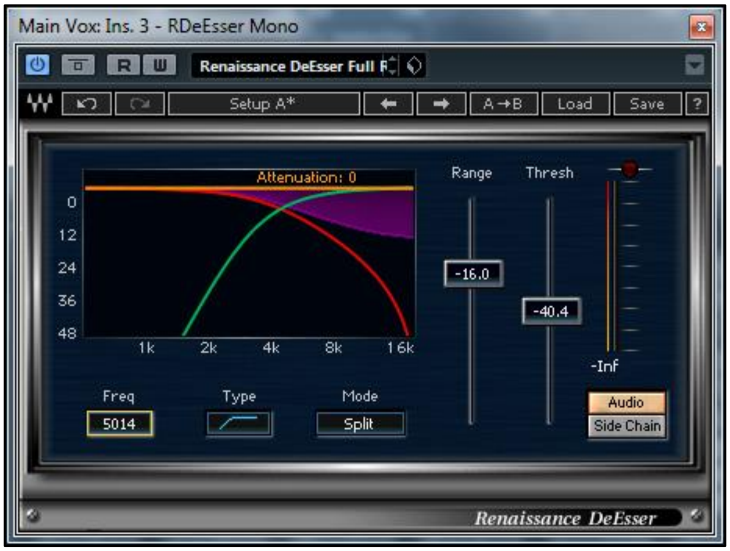
Distortion

Technically this can be considered a part of 'saturation', but it really depends on how far you want to push it. Most vocals which compete for dominance amongst strong midrange elements, such as distorted guitars, will tend to like some form of distortion, both to blend them into the mids and also to stand out a little. Traditionally this was done during tracking or mixing by cranking the console mic preamp right into the red, until it gave you the right amount of grit.

These days the workflows are all over the place, so what you need to establish is what distortion tool is the one for you. You can use your transformer saturator plug-in and send it into overdrive, or you can use a dedicated distortion box. Some use tube screamers, others use guitar amps, some still crank mic preamps into the red. It all depends on what sort of craziness fits you best, and beyond that, the mix.

In order to get the grit from distortion while still maintaining intelligibility it can help to run the distortion processing in parallel. This way you can be very heavy-handed with the gain, and simply back it off on the dry/wet blend until the vocal sits right.

De-essing



Provided the vocal has been tracked right, and you've hit it with a bit of saturation to help it along, you should find that your need to de-ess should be fairly minimal.

You can use a dedicated de-esser tool, a multi-band compressor, an EQ, or side-chain an EQ into your console strip's compressor, boosting the offending frequency, so that the compressor will clamp down on it. Use whichever feels best to you. Generally, a good dedicated de-esser tool is worth its weight in gold.

The center frequency needing de-essing is very dependent upon the vocal chain and vocal itself. Generally you're looking anywhere between 4kHz to 8kHz. A 'safe' bet to catch everything is to set a shelving-type de-esser at a starting frequency of around 5kHz. This may lead to a slightly dull vocal, which you can later make up for with high-shelving EQ.

Mixing

EQ

Mixing vocals isn't so much about getting them to fit into the song as it is about getting the song to fit around them. Remember this, because it's very important: *You don't want to be mangling your vocals with too much equalization.* If you are, chances are there is either something very wrong with the vocals themselves or the rest of your mix.

Start by reaching for your best EQ. I would recommend that if you were going to use hardware equalization anywhere that it at least be here. Vocals are all about the air you get from strategic high-end boosting, so you will want a colorful, euphonic EQ to form the basis of that. Since most vocal EQ will generally be quite broad, you shouldn't need to worry about running out of bands for surgery. Begin with a high-pass filter. You can

generally set this anywhere between 60Hz and 150Hz, depending on how much low-end content is useful to the mix. The general rule is that the busier the mix is, the less excess junk you want flying around below the vocal.

Stage your EQ so that it runs into your compression chain. Try to always EQ into the compressors if possible, because this allows you to crank more highs in to the vocal. The compressors will react to this, suck down it, thus allowing you to get that floating bed of 'air' above the vocal. Since vocal processing is quite broad, we can break our EQ'ing down to 4 bands, which should also coincide nicely with the number of bands on our expensive color-EQ!

You might first want to listen to the mids, and make sure there isn't any pervading flatness. Many vocal chains will generally have a bit of mud between 300Hz and 900Hz. Sweep it around, find your spot (if there is one) and subdue it only as much as you have to. Play around with the Q a bit, as we're trying to get this done in as few moves as possible.

Now get your high shelf out and crank some highs in there. This is the fun part, and if the job up until this point has been done right you should be hearing the vocals come alive and start to sparkle. If the 'esses' are starting to pop out again, but the vocal is otherwise not feeling very bright or vibrant, try clamping down a bit harder with your de-esser chain, and then get a bit more highs into the shelf. Work the two against each other until you've found just the right balance for the vocal.

Generally 8kHz to 10kHz make good starting points for the shelf. The severity of the boost will really depend on the vocal, your EQ, and the sort of tone you're searching for. General tip: if you really want the sound of the EQ high-boost in there, use some heavy 15 IPS tape saturation, console saturation and perhaps 'vintage' flavored mics, preamps or compressors to bleed the highs off, and go heavy on the de-esser. All these things together should give you a fairly blanketed vocal, which is ripe for diming high-end into. Your goal is to subdue the abrasive, harsh highs, which may be endemic to the singer's voice, the recording chain and their proximity to the mic, in order to be able to emphasize the highs that you want with your processing chain.

Depending on the microphone's tendency toward the proximity effect, you might find yourself needing to attenuate the lows in order to stop the vocal from clashing with the lower elements of the track, such as the bass guitar, or snare thud. Reducing this just enough is an exercise in maximizing headroom, and will let the vocal float 'on top' of the track easier since the plosives won't be pushing it down below the music.

A low-shelf here is fairly common. Try setting it anywhere between 100Hz and 200Hz. Jog around the depth of the cut and the band until it sounds right in context with the mix.

On a standard 4-band parametric EQ, we've now got only 1 band left. Use this one to take a look at the mids/high mids of the track. Some like to boost 4kHz for a bit more bite and cut, others like to boost between 1kHz and 2kHz for more of a 'telephonic' effect. Jog the band around and do what's right for your vocal. However, be cautious of

boosting the 'presence' frequencies too much, especially if you plan on compressing heavily later, as they will naturally be brought forward. You may even find yourself needing to cut the 'presence' frequencies in order for the vocal to nestle itself into the mix, rather than sounding really uncomfortable and 'on top' of everything else. The latter is generally an issue when staging many compressors, and amounting to a very high overall gain reduction.

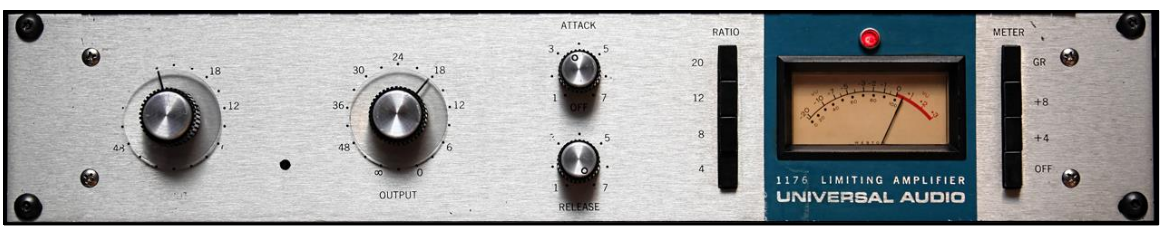
Spectrally, if the vocal still doesn't sound right, then it may be a case of needing to pull up your workhorse surgical EQ and taking a look closer. Sometimes singers, mics, or vocal chain combinations have anomalies, which we need to treat a bit more surgically than we otherwise would like. The key thing to bear in mind is to be minimal here, as the more EQ you do, the more you inherently degrade the fabric of the vocal. These anomalies could be all over the spectrum, depending on the track. The workhorse method of jogging a boost around until a frequency peaks out in a very uncomfortable manner is a good way of finding your starting points here.

If after all of this the vocal somehow still sounds too bright or abrasive, then don't be afraid to give it a gentle low-pass filter. Some vocals like to be mid-concentrated and there is nothing wrong with channeling them there, provided the mix calls for it.

Compression

Ideally, we would have been EQ'ing into our compressor chain from the start. Once you work out your ideal vocal compressors and their 'sweet spots', this will become second nature and you will work between the two seamlessly.

The name of the game with vocals, much as it is with bass guitar, is 'staged compression'.



It's not unusual to hit modern vocals with up to 4 different compressors. The key being that each plays its own part in the overall equation, and stops the others from being overloaded as they would be when working individually. What you're effectively trying to do is chase the vocal tone with the compressors, and severely reduce their dynamic range so that they become controllable from a mix perspective. The dynamic movement is then automated back in manually over the course of the mix process. If the resulting waveform looks like a brick, then you're generally on the right track.

A classic starting point here is the optical into FET combination. Hitting the vocal with a gentle optical compressor for a substantial amount of leveling, and then following up with an 1176-style FET compressor to catch the peaks. This creates a very natural and very analogue sounding vocal. Depending on how you balance between the two you can really control the range of the vocal and how it sits in respect to the mix. You can also

use the optical compressor for a little bit of leveling to get it started, and then use the FET for the majority of the grunt work if you're after something more aggressive.

If you're looking for the classic LA-2/1176 combo then you want your optical compressor to have slow attack and release. It has to sound gentle so that it can cope with the brunt of the work. This works well with pop or styles where the vocal itself has to sound gentle. The 1176 should be set to work at extremes, so fast attack and release, and either 4:1 or 8:1 ratio, shaving off only the peaks.

If you're looking for a more aggressive sound, consider swapping out the optical compressor for a workhorse VCA compressor, like the Distressor. One of its defining characteristics on vocals is its ability to absolutely pin the vocal with gain reduction, while still retaining the illusion of transparency. Setting the attack to medium, and release fairly fast with a ratio anywhere between 2:1 and 4:1 tends to be a great starting point. Gain reduction should be set to taste, and for the project at hand. The distortion modes are also great to experiment with. Follow this up with an 1176-style FET compressor set with a moderate attack (which is still ridiculously fast on an 1176), and a fast release. This combination is absolute gold, and should be able to satisfy 90% of your compression needs for a vocal.

Follow all this up with a brick wall limiter in order to stop errant peaks shooting through the chain and messing up your mix. Be very careful here and use only as much as you feel necessary, as you can really make a vocal sound thin and distorted unless you're careful.

Provided a compressor was used to control the vocal during tracking, you've now totaled up to four dynamics processors (five if you count the de-esser). This should be plenty enough to get the vocal under control and give you the tone shaping you need to make it sit within the mix. The interaction between EQ and compression is paramount, so make sure to constantly balance them against each other and ensure you've got the right staging for your vocal.

Multi-band on vox?

Multi-band compression on vocals should generally be viewed as a corrective measure. Think of using it to de-ess or de-plosive a vocal. It can control sudden bursts of air into the mic, which might result in a woof of mid-bass and low mids, and it can control abrasive sibilance in the higher frequencies without eating into the tone of the vocals. Otherwise, the multi-band compressor can be used in situations where you are struggling to get the vocal sufficiently airy. It can be set to compress only the 'super high' frequencies and boost them with the make-up gain. The effect of this is a super-compressed, always-present top-end which can at times be just what your vocal needs in order to sound alive.'

Creating your donut hole

The 2kHz to 5kHz range within your mix should be unobstructed and waiting for the vocals to fill it. It's almost too easy to get overheads and guitar tracks which will just eat up all this high-mid room. This is where the vocals live, after all, and if there is ever a

clash, they should win.

Always aim to build the mix around the vocal, so that you can minimize the amount of pushing you need to do on the vocal in the high-mid frequencies. Ultimately the more ‘pushed’ the vocal sounds there, the more unpleasant it is to listen to for prolonged periods of time. People are very sensitive to the 3kHz range, so you need to go easy on how much information you funnel down through there.

Effects and other trickery

An un-effected vocal will generally sound quite isolated and thin, unless it’s been captured in a live room from a distance where the room ambiance is a factor. This is quite uncommon in this age of dead vocal booths, so it’s down to us to create our flavor of ambiance and presence for the mix.

Widening

A good starting point is to use a micro shifter/stereo widener in order to give an impression of width and size to an otherwise mono vocal. This is a tool that will take the vocal out to the sides, delay them and pitch shift them slightly in order to create a ‘fake stereo’ effect. Use this in small dabs, as over-doing it might give your audience headaches. The idea is to create the feeling of subtle early reflections to ‘enlarge’ the presence of the vocal within the mix.

Room reverb

It’s a good idea to use a small or medium room reverb for the first reflections. This serves a similar purpose to the stereo widener, albeit a little less fake. You’re still trying to create density for your vocal at this point. If you’re working on a ‘larger sounding’ production, you can interchange this with a hall reverb, and use a plate as a follow-up for extra density and decay.

Follow-up reverb

This can be anything from plate to chamber to hall to ambient reverb of some variety. The idea is to use this for the tail of the vocal, and give it a greater sense of depth within the mix. Try using it in mono, in order to isolate the vocal ambiance within the mix and make it sound more surreal.

Delay

This is one where you can really go to town. First make sure that the delay follows the tempo of the song, and then set to the beat division that you want it delaying to. Quarter notes are a good starting point for day-to-day work, but some prefer the sound of quicker slap-back delays for certain styles of music. Next, work out the sort of saturation you want on the delay. Tape style, or telephonic delay tend to work really well because they subdue the highs and make the delays blend into the mix rather than stomping on top of the main vocal line.

Parallel compression?

While this is something I wouldn’t commonly employ on a day-to-day basis, it can help

with a vocal that's starting off quite thin. The parallel compression can help inject some body back into it, and potentially give you a better starting point for the mix.

Automate, automate, automate, and then do it some more

The whole idea behind quashing all the dynamic range in the vocal is so we can automate the movement back in ourselves. Some ride the volume fader on the vocal all the way throughout the song, choosing how forward or back each inflection should be. This is quite valid, but it's also possible to work in sections too, especially if the music is somewhat un-dynamic, let's say death metal rather than pop rock, where the song structures aren't intended in the commercial 'build toward the chorus' way. Take a few practice runs. See what fader moves you can do to make the vocal 'pop' more. Highlight it in sections where it needs highlighting; take it back in sections where you want more distance and space.

Automating delay and reverb throws is also a great method of getting some movement and flow into the song. There are usually lines at the end of phrases, which are very suited to this sort of thing. Keep your ear out for them. Big deliveries and big finishing lines usually call for a dollop of extra delay or reverb. Try combining them with a fader dip on your vocal track in order to make the vocal feel like its fading into the distance and getting larger as it goes.

Try keeping an FX delay on hand set to 2nd notes, or whatever else works for the song, and use it for your bigger delay throws. Sometimes a Ping-Pong or wacky panning style delay can do the trick too.

Some people also feel the need to automate EQ between sections in order to keep a vocal sounding consistent. While I've personally never found a situation where simple tone-chasing and level automation didn't work, this can also be accepted as a valid avenue to pursue if your mixing style calls for it.

Backing vocals

As one might imagine, the principles behind mixing backing vocals don't differ all too greatly from mixing lead vocals. The additional concern that they carry is their need to augment the density of the song, without actually drowning out the lead vocal entirely (unless, of course, that's the desired effect).

A great way to achieve this is to use ‘gentle’ compressors on the backing vocals. Optical or tube-optical designs with an inherently warm character can be great for large amounts of gain reduction here. You can get quite extreme with backing vocals, as you want their footprint within the mix to be relatively minimal. If you have a good compressor on hand, feel free to pin them, and consider following that up with a limiter-like compressor in order to control any leftover, errant transients.

Regarding EQ, consider shaping them in a similar way to the lead vocals, but then sucking out a fair degree of content between 3kHz to 4kHz. This will create ‘nest’ for the lead vocals to sit right among the backings, yet still be distinguishable between all the layers.

Backing Vocals tend to love widening chorus effects. If you're struggling to get them sounding textured, wide or thick enough, then add a splash of slow-sweeping, shallow stereo chorus and hear them come alive in the mix.

Frequency spectrum summary

0Hz to 150Hz:	Can generally be filtered without issue.
200Hz to 400Hz:	Prone to containing ‘mud’ from the mic and proximity effect. Consider attenuating in order to open a vocal up.
500Hz to 800Hz:	Very common area to need wide reduction. Will help the vocal sound more ‘3D’.
1kHz to 2kHz:	Telephonic mid-range. Can be useful to boost in order to cut through very dense mixes, but will make the vocal sound somewhat unimpressive by itself.
3kHz to 5kHz:	‘Presence’ range. This is where the vocal lives in your mix. Be very mindful of what is contesting this region. If the vocal needs to be boosted too much here, other elements in your mix may be eating into it too much. Decrease this region to increase the ‘depth’ of the vocal, at the expense of drowning it out in the mix.
5kHz to 8kHz:	Commonly contains sibilance.
8kHz+:	‘Air’ frequencies. Can be boosted to add a nice sense of liveliness to the vocal.

Chapter 7: Controlling your wood (mixing acoustic instruments)

One of the trickiest things about mixing into arrangements that are dense with saturated, distorted, spectrum-thieving elements is the act of trying to squeeze entirely acoustic instruments through all that. Your real estate is quite small, but you still have to make the most of it.

Chances are you're dealing with something made of wood, with pieces of metal drilled into it and other metal or nylon things resonating in order to create the sound. This is somehow meant to cut through a screaming, broadband wall of distorted guitar? It isn't, but at some point someone decided it could, and that it would sound good, and another person agreed. Now you've got a ton of clients who expect it to be done. So buckle up and read on.

Acoustic instruments such as piano, cello, acoustic guitar, upright bass etc. tend to be very rich and complex with their low-mids. While this can lend a lot of warmth to their inherent sound, it also creates the first and biggest stumbling block to making them work in concert with busy arrangements.

Mixing acoustic instruments takes from techniques used to process both percussive elements and those used to process harmonic ones. The basic EQ curve you're bound to come up with on these instruments tends to approximate those you'd be putting on your drums. The compression approach is unique, because over-compression is extremely audible on these instruments, and the sweet spot is extremely small.

Saturation

Acoustic instruments tend to respond very well to saturation, as it's a more transparent sort of compression. Console saturation is a staple, and tape saturation can really help out on instruments that are inherently abrasive. Gentle sounding saturation is the way to go, as it will help unify and glue the sound together before you've moved onto doing anything else.

Automation

Since acoustic instruments tend to respond so poorly to large amounts of compression, it's very important to get familiar with riding the volume fader through the sections. Ultimately using the compressor as a tone shaping tool and using the fader to control the dynamics will lead to much more transparent and pleasing results in this case.

Mixing

There are a lot of instruments out in the world. We could cover the most popular ones with specific approaches, but that would ultimately be counterproductive. It makes a lot more sense to equip you with the basic tools to tackle any which come your way.

There are two ways to approach this, and they are dictated entirely by the form of song you're working on. The first implies that you're trying to squeeze these instruments over the top of a dense arrangement, usually as flourishes to help add an extra density and air to the song. The second implies that these instruments are holding down the bulk of the frequency real estate and perhaps even functioning as the fundamental bass elements of the track. The former could be any modern pop, rock or metal arrangement and the latter could be a ballad or any form of music driven by acoustic instruments.

Dense approach

The first word to take home here is 'downsizing'. Whether we're mixing a piano, acoustic guitar or whatever else that's going to be running over the song, we need to make sure that it leaves space for the other instruments, which form the foundation of the mix. The biggest areas you will generally find yourself looking at is conflict with the bass guitar in the mid-bass frequencies, low-mid mud that will impact on the clarity of the mix as a whole & getting the highs to cut whilst still leaving the presence region subdued enough for the vocals to sound effortless over them.

Try our trusted method of staging the EQ into compression. This way the compressor will constrain any substantial boosts you may be inclined to do, and give you a smoother end result. You will want to use a broad, sculpting EQ to start with. An optical compressor in the vein of an LA-3 or LA-2 is a natural follow-up.

Start with filtering. If it's a very busy arrangement and you only expect the instrument to be cutting through the highs then you can be quite brutal here. Anywhere from 60 to 180Hz ought to do for your high-pass.

After this, look at the low mids. This is likely where you will do the bulk of your cutting. If the instrument sounds very congested and tubby, look between the 200Hz to 500Hz region. A strategic, wide cut at the right place here should get you well on your way.

Next, you will find that many acoustic instruments that have body in their tone will have notoriously muddy content between around 160 and 250Hz. This is a very dangerous crossover area between the mid-bass and low-mids. Too much content here and you will just create booming and mud. Too little, and it will sound extremely thin. Many acoustic instruments need a substantial cut in this area in order to tighten up their low-end response. Where precisely, depends on the instrument's construction and dimensions. If cutting this area leaves the instrument thin, you can warm the bottom-up with a low-shelf set anywhere around 100Hz. This will generally give you more quality and deep low-end rather than the tubby variety.

Many instruments will have resonant nodes or string noise between the 2kHz to 4kHz area. Pianos are particularly notorious for commonly having content around 2kHz. This area absolutely **must** be controlled in order for any vocals to sit on top naturally. The best way is to run the vocals and instrument at the same time, and EQ around the high mids until you've taken out the offending range.

After this you can commonly give a fairly substantial boost to the high frequencies in order to increase air and help it cut through the dense arrangement. The best way is to run the instrument alongside the mix and boost until it sounds right in context. Outside of context it will likely sound quite thin and harsh, but this is just the way mixing goes.

The compression element is fairly straightforward. The easiest way to get an acoustic element sitting snugly in a mix is to run it into a basic optical compressor like the LA-2 or LA-3. Since these have no attack and release parameters, it ends up being a no brainer approach where you simply dial in your desired amount of compression until things get a bit too much.

Whichever compressor you opt for, the key is to not set the attack too fast. There is a percussive element to these instruments which is necessary in order to work with the mix. A gradual release tends to be crucial for them to remain sounding natural even when clamped down with compression.

At this point the instrument should be EQ'd and compressed quite well. One thing you might notice is that it sounds a bit 'pokey'. Some transients are really overshooting the body of the tone and creating undesirable peaks within the mix. The solution here is much the same as drums. Pull up a limiter and clamp down on those errant transients until they sound right.

If you're really having trouble getting the low-end in check, then it might be time to bust out the multi-band compressor. This tool will provide a great way to subdue any muddy resonances down below, without eating into the body of the instrument entirely. It can be absolutely invaluable when sculpting an acoustic instrument to work in concert with a bass guitar.

Once again, it's worth stressing to start off by going easy on the compression and limiting at the start. It is very easy to destroy the beauty of an acoustic instrument's sound with too much compression. Sometimes in a very dense arrangement you absolutely have to clamp down, but in any circumstances where that instrument takes the forefront, you want to avoid pumping and tearing artifacts at all costs.

Sparse approach

This approach assumes that you are working on a song where the acoustic instrument is taking the bulk of the space in the mix. Whether it be a ballad, singer-songwriter, country or whatever else song, the principles remain quite similar to the approach we detailed above. With one exception - being less extreme.

It doesn't matter what type of song you're mixing. An acoustic instrument will still have its resonant and muddy areas in the same places. The only difference is that you now

have the real estate to keep more of that in place, and keep the instrument sounding larger.

One of the most important things to identify is which instrument will contribute to the bulk of your low-end. Sometimes when there's no bass guitar, fretless bass, upright bass, or any permutation of bass-specific instrument, you have to double-back one of your other focal instruments in this role. Sometimes that may be a piano, other times it may be an acoustic guitar, and other times still it might be something much more exotic. Once you've identified this foundation-block of your mix, you can then start to build the other instruments around it, thinning them out so that they don't conflict in the crucial low frequencies.

Optical compression is still your go-to flavor here. You're trying to avoid any sucking, pumping or tearing artifacts at all cost, so the gentle nature of optical or tube-optical compression can really help you get there. Slower attacks, slower releases and smaller ratios are the name of the game. If you find you can't get enough dynamic control using the compressors, then simply automate the volume through the sections. The song will breathe a lot easier and sound a lot better as a result.

Effects

In busy arrangements your 3 weapons of choice for acoustic instruments will be chorus (for subtle thickening and widening), delay (rarely, and being quite careful with it) and reverb (commonplace). The reverb can generally be anything from a room to a chamber to a hall, depending on the song. A bit of chorus is nice to have in general because of the wider and more textured sound it creates. Delay is somewhat rare, but can help in percussive or deep sections that need a bit of bounce-back.

Frequency spectrum summary

0Hz to 60Hz:	Generally safe to filter out.
60Hz to 150Hz:	The bulk of the low-end content. Can be boosted in order to add weight to the instrument.
150Hz to 250Hz:	Tubby resonances are generally found here. Subdue heavily if need be.
250Hz to 500Hz:	Low midrange mud. You will generally cut deep and wide here.
500Hz to 800Hz:	The ‘flat’, ‘2 dimensional’ range. If an instrument sounds papery, look at this region.
2kHz to 4kHz:	‘Presence’ range. Make sure this is controlled in order for vocals to sit well.
8kHz+:	Boost here in order to add air, cut and sheen. In a dense mix you sometimes need more than sounds right when soloed.

Chapter 8: Downsizing synthesizers

Synthesis, in some shape or form, has pervaded most popular music since somebody in the 70s took a bit too much acid, grew a fondness for tights and thought that Minimoog solos were a good idea. Mixing synthesizers among 'real' instruments has been a mixing staple for a long time. Obviously the sounds that synthesizers can make range so greatly, that there is no possible way to cover them all specifically. So by nature, this has to be the most general of all the mixing chapters. Synths can throw curve balls at you and really test your creativity and vision for a mix.

While synths can be one of the most non-standard elements to mix, they can also be the most forgiving, simply because many patches come out of the devices just about ready to go. Most of the time they just need downsizing in order to sit with other instruments effectively. It isn't a matter of having to really sculpt the sound as one might need to do with a raw drum kit or bass guitar. Heavy filtering is quite common practice with synths, and usually quite necessary in order to get them functioning well with each other. The first thing to do with a synthesizer track is determine its role within the mix. If it's a sub-bass, then you may want to treat it as you would a bass instrument. If it's a lead synth then you might want to treat it as you would a lead guitar. If it's a pad then you may want to treat it as an ambient element. You have to isolate where it's going to sit in the sound field and how.

The most dangerous of synth type sounds tend to be the pads. The problem with many pads is that they want to eat up the entire spectrum. You have to be selective about where to downsize them and where to let them cut, in order to not lose your entire mix the moment you un-mute them. It helps to process them in context with the mix, be judicious with your filtering, and not afraid to cut away drastically at regions of the spectrum they try to dominate.

Contrary to what you might think, synths tend to be quite responsive to heavy saturation, distortion and compression. These elements can really help them come alive and cut straight through the mix like a hot knife. Experimentation is the name of the game there. Tape saturation can take some of the edge off digital synths, while hard compression and distortion can really make them pop out of the mix.

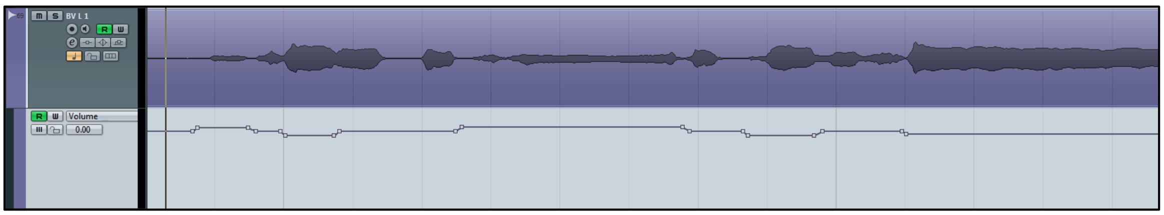
Synths can be a playground for chorus, phasing, delay, reverb or just about any other effect you can imagine. Most take remarkably well to chorus, or stereo widening of any kind. Ping pong delay, slap back delay, long delay, and all sorts of reverbs whether realistic or fake can work depending on the song. It's really an area where you look at the song, figure out what it needs and then make it happen with the synths.

Chapter 9: Flying faders and level rides

The art of mixing exists in order to create a bridge between the song and the listener. The mix is the delivery system, or the vessel of the music. There are a few key areas where mixing overlaps into actually augmenting the song itself, and automation is undoubtedly one of them.

Automation is very common in top-level commercial mixes. It helps create a sense of dynamic movement even after all the tracks have been potentially squashed to death. You can automate anything from track volume to delay throws to FX to whatever else you can imagine might benefit from it.

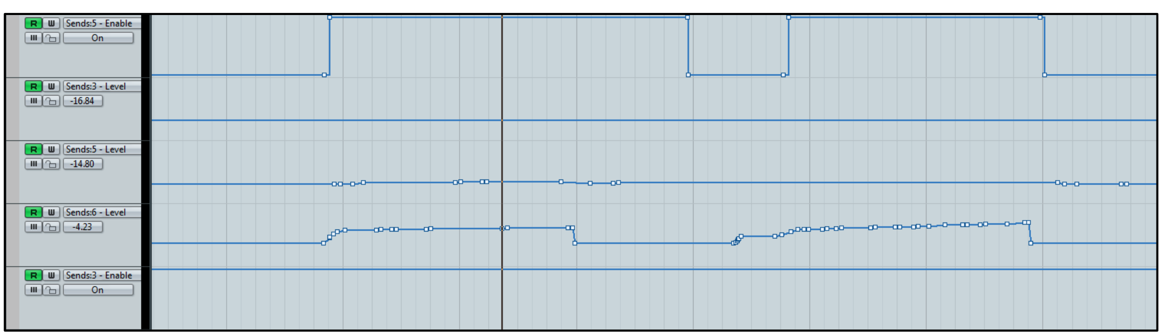
Corrective automation



Corrective automation is the type that needs to be done before any further steps can be taken. It's generally confined to level and EQ automation. An example may be automating the volume of a vocal line before it hits the compressor chain, in order to reduce the workload on the compressors, and stop them overloading. Another example could be EQ'ing various vocal passages differently for a vocalist who is 'lively' with the mic and frequently changes the angle and distance from which they sing into it. Another could be dipping the low-end of a bass drum as it bursts into a quick double-kick passage, so that the low-end of the mix does not overload.

Corrective automation can generally be avoided if the tracking has been performed well – but there will usually be one or two things within a mix which require being controlled before moving on to the next stage.

Creative automation

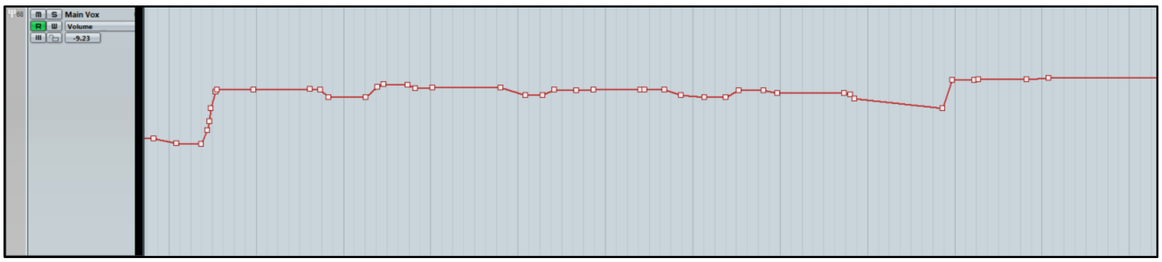


Creative automation can be seen as one of the most enjoyable aspects of mix engineering. When the mix is at a point where this kind of automation can be done, it usually implies that all the balances are in check and all that's left is the creative input from you, the mix engineer.

An example of creative automation could be setting up delay and reverb throws for the lead vocal line on every ‘large’ tail-off. Another example could be manually riding the levels of the drums, guitars, bass & vocals between song sections in order to give the song more dynamic impact. Another example could be automating a Flanger effect, a filter sweep, or a pan sweep for an effected break in the song.

Creative automation encompasses many different elements of mixing. Effectively any time one is being creative in the mix, and recording their dynamic input to either the volume level, pan, sends or their processor parameters can be considered a part of creative automation.

Volume automation



When we think of automating something, the volume of the tracks is undoubtedly the main use of the technique that comes to mind, and rightly so, because it’s one of the most important.

Volume automation helps the track bring out a dynamic feel, and recreates the sort of movement that may be present if a band were to play the song live. The amount of automation you do here would depend heavily on the song, but also partially on your own inclination.

If it’s a very chorus-oriented song, then chances are you will be automating a lot between sections in order to help the song build toward the chorus. You might be inclined to reduce the level of all the instruments on all the prior sections, incrementally increasing them as you get closer to the chorus itself. This can take a bit of time, but it’s an investment worth making, and necessary to get away from an ultra-modernized, sterile feel to the music.

Automating between sections can be a nice, broad way of sculpting the song, but many people also like to automate within sections. This really depends on how expressive and nuanced the song is, and how much of that can be highlighted with volume moves. Even in rudimentary rock songs, some people like to automate the vocals by hand. In a commercial arrangement, the vocals are the focal point and it makes sense to give them the royal treatment. If they are slammed so they have virtually no dynamic range left they can sound very boring unless you manually move them around a bit to punctuate the dynamic feel of the song.

If there is a short section where you want to highlight an instrument or performance, don’t be afraid to make it ‘too loud’. This is the best way of catching the listener’s attention, and if it’s only in passing, they won’t think much of it. Mission accomplished.

Whatever you can't do with track or bus automation you can finish off by riding the fader on the master bus. Don't be afraid to swing this one around a bit between sections. It can help give the song an added push or pull during crucial moments.

If you're very heavy handed on your master bus compression, a lot of volume automation is not only a bonus, but also an absolute necessity. It's needed to help the song sound coherent and avoid the quiet sections ending up louder than choruses.

Pan automation

This is a very straightforward technique. It essentially involves altering the panning of a part as it's playing. It can be used to create some nifty effects, but is very easily over-used. There are only so many times something can go from left to right, or vice versa before the listener starts plotting to kill you.

Delay automation

Delay throws are a great, classic technique to help emphasize and extend important words or notes. A good starting point is to set a delay to 2^{nd} notes and send to it from a track every time a 'big' note comes along. Whether this is a sustained vocal note, or a sustained lead guitar, try it out and listen to how it gels with the song. It won't always work, but when it does, it helps add a subtle excitement to the flow of the song.

Another option is to automate the level of your main delay (hypothetically, let's say a quarter-note delay which is constantly engaged). This can help emphasize the depth and 'surrealness' of a vocal for sections where the bigger 2^{nd}, or 1^{st} note delays are overkill.

Experiment with ping-pong delays, stereo delays and effected telephonic, dull, tube or otherwise 'weird' sounding delays for extra fun. These can all help merge the song with the production and bring the whole thing to life.

Reverb automation

The exact same principles apply here as with delay automation. The added bonus is that this can be done during busy passages and won't overlap or intrude on the subsequent line as much. Simply set up a big hall or big plate reverb to help carry a note, and then send to it whenever feels appropriate. This can help increase the perception of a 'loud' note, especially with vocals, as the reverb to dry ratio changes and the word immediately becomes highlighted from the rest of the mix.

Another good use for reverb automation is to 'enlarge' a drum kit, or snare in particular for chorus sections. You can use it to have a close, intimate drum sound on cleaner verses, but then really open up in the big busy moments to increase the sustain and air on the kit. Be creative, use this anywhere you deem appropriate, and help bring the vibe of the song out.

FX automation

There are an absolute ton of elements to automate within a mix. How effect and automation heavy you want to go depends both on the song and your style. You can automate telephonic effects to certain instruments, time-based effects like flanging or

phasing during transitions, filter sweeps, or just about anything else your imagination tells you can be done. All of these can be used to create ear candy and break a song up from sounding too much the same.

So, as always, once you establish the fundamental uses of the technique, go out and experiment. Automate everything until you find your sound and your style.

Chapter 10: Using the 3rd dimension

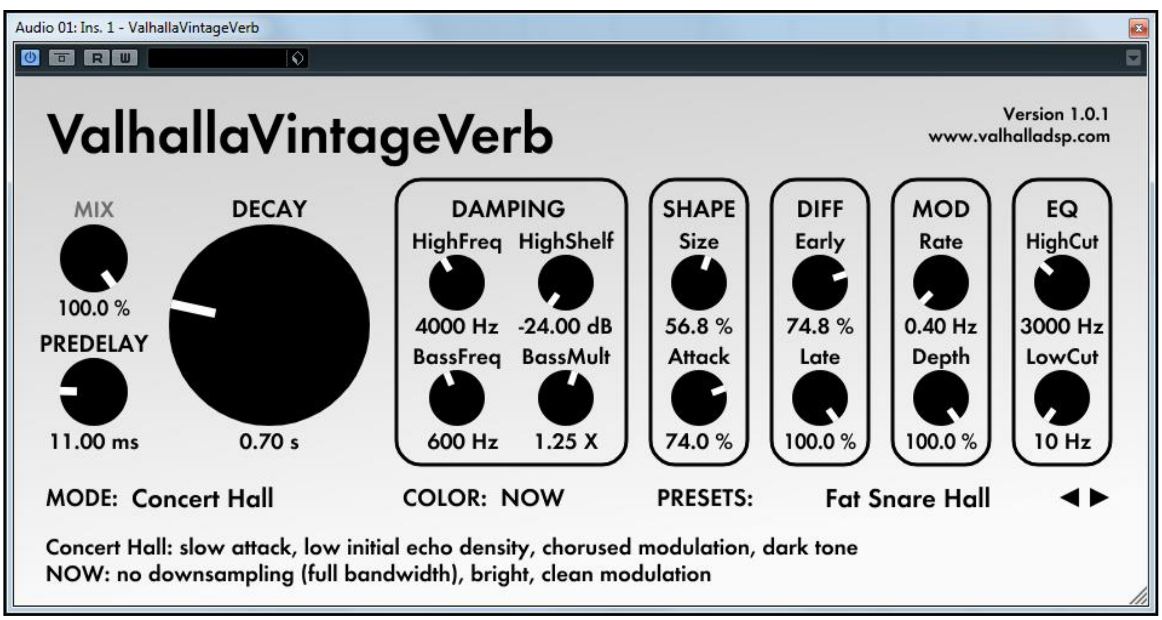
Most people visualize music coming out of their speakers across a singular plane. Left to right, perhaps with a touch of frequency-dependent vertical element thrown in, depending on how the woofers and tweeters are arrayed. Yet there is a key, illusory plane which allows us to establish the size and depth of the record. It doesn't actually exist, but a well-mixed song will convince the listener that it does all the same. The elusive 'back to front'.

We've already established that much of the perceptible 'size' of a mix comes from clever use of equalization and compression. Removing obstructions across key frequency bands, and using each mix element to effect creates the aural equivalent of completing a jigsaw puzzle right in front of the listener. Yet why would we stop there? We can turn that static, 2-dimensional image into a full blown hologram.

The key lies in effects processing. The subtle use of time, phase and frequency variable effects to make the music feel as if it is somehow coming from 'outside' of the speakers.

Reverb

The singularly most common effect used across all mixes in history is reverb. So much so, in fact, that this statement is even backed up by the Institution for Audio Facts Nobody Cares About. Now that you are aware of this morsel of absolute truth, it should become apparent that mastering the use of reverb is key to opening up the 3rd dimension in your mixes.



Reverb is crucial to framing the sense of space around each mix element. Even when it isn't perceived outright as reverb, it can still add a subtle yet pronounced sense of belonging to a track. It is perhaps the most universal effect in terms of its application. A certain degree of reverb can be used on just about any mix element. In fact, it's easier to talk about the mix elements that reverb *wouldn't* work with.

In general, you want to discern which elements of your mix form the core, rhythmic, driving foundation. In many cases these are the bass drum, bass guitar and rhythm guitars. Any element which has a substantial amount of low-end energy that is rhythmic or driving in nature is safe to stick into the 'minimal to no reverb' category. Adding reverb to low-end dominant tracks is a great way to smear one of the key foundations of your mix, so be wary.

Just about anything else is fair game. Snares, toms, room tracks, cymbals, pianos, acoustics, strings, vocals etc. etc. The biggest unknown quantity for you will be discerning 'what type of reverb, where, and how much'. So without further ado, let us explore some common reverb algorithms and their potential applications for your mix.

Room

Room algorithms are commonly defined by a fairly short decay time, strong early echo density, and substantial diffusion. The whole idea being to add a sense of space and early reflection density, without it being audibly perceptible as a fake 'reverb' effect.

The most common application for this type of reverb is for any track where you might want to add a sense of girth and space, yet without overstating it with a long tail. It works especially well on quick-firing rhythmic tracks where you don't want a long tail to smear the punch.

The most common elements to use room reverb on are drums and vocals. Dialed subtly to augment the natural room tone of the drums with an artificial sense of weight and depth in the former case, and dialed to create the impression of the singer being in a small to medium space in the latter. I often find that dense tiled, or wooden room type presets form a good starting point for drums, and that live, diffused studio type presets form a good starting point for vocals.

The way in which vocals are commonly recorded results in a highly unnatural and 'dry' sound. The human voice is not often perceived in a dead environment, a few centimeters from the singer's mouth. Therefore, room reverb is a great method to inject some reality and space back into the track, and allow it to blend with the instrumentation instead of sitting squarely atop it.

Other applications are strictly to-taste, but you may want to try experimenting with using room reverbs on any acoustic elements to enhance their presence. It can also work wonders to merge synths with an organic band arrangement, by providing a sense of context to otherwise very disconnected and unnatural sounds. Try to think of the room reverb as a 'girth, with a touch of air' slider for your mix elements.

Chamber

Chamber reverbs are generally characterized by strong diffusion and echo density. The idea being to pack the reverb with a lot of density, yet in regard to tonal coloration, keep it more transparent than plate algorithms.

This is a tough algorithm type to speak of, as I personally don't use it very often at all.

The general impression it gives is that its utility lies in making tracks more lively and exciting. For instance, it could be used on lead guitar or solo tracks in order to create a dense sense of space and energy around them.

Likewise, it could be used on string or orchestral ensembles to achieve a similar, energetic, yet slightly more intimate effect than one might get with a full-blown concert hall algorithm.

Hall

The hall algorithm is the mainstay of reverb processing. It's as close as you're going to get to a 'vanilla' flavor. When people think of artificial reverb, odds are they're thinking of the hall algorithm.

It is commonly characterized by a deep, lush, modulated tail, highly variable echo density and diffusion, and a large dimensional presence within the mix. Its application is boundless. Just about any track can benefit from a degree of hall reverb, whether it be a thunderous snare, a concert ensemble, lead vocal track, or an EDM synth.

This algorithm provides that richly modulated decay which can create a tremendous sense of space for the right mix element. Hall reverbs commonly shine at 1 second and beyond decay times. They can even be used to create artificial, soundscape-like sounds by being pushed to extremes, such as 7 seconds and beyond. Yet, for the majority of your processing, attempt to keep the decay time between 1.5 and 3 seconds, as this is the most natural range.

Pre-Delay can be a very important control for the hall algorithm, as it determines how long the gap between the dry sound and the onslaught of reverberation is. This can help space out the dry source from the reverb, and create the perception of even greater size. If you can imagine standing next to a snare drum as it's hit in a tremendous hall, obviously your ears are going to receive the dry sound first, before the echo makes its way back from the bounds of the establishment. Pre-Delay can be set to taste, yet I don't often find myself using it beyond 30 milliseconds.

Damping is quite important as the reverb tail is so audible. You might often find yourself damping the higher frequencies quite substantially and boosting the lower, in order to create a greater sense of girth and depth to the track. Long, bright hall sounds can be quite unnatural to the ear.

Modulation is often provided as a control which can be tweaked on a basic level by most reverbs. While this isn't an essential control to master, it can help in sculpting the lushness of the reverb tail to fit your mix.

Medium hall presets often form a good foundation for use on drums and vocals. Concert, or large ambient hall presets are often great starting points for orchestration or synths.

Plate

Plate reverb brings color to your mix. Think of it as that quirky uncle you had as a kid, who always seemed to drink a little too much at stoic family gatherings, injecting a bit of

fun into the experience.

Plates are commonly characterized by a bright attack, very high echo density and diffusion, with a lush, unnatural modulation. It is important to understand that plates aren't designed to sound realistic or transparent. They are designed to sound lively.

This makes them a great candidate on drums and vocals. In particular snare drums and lead vocals. In fact, mono elements often work best with plate algorithms, and at times the plate algorithm itself can work best when set to mono. It can create a surreal, wrapped sense of space around a key mix element, and identify it from the rest of the mix. This can work well in drawing special attention to snares and lead vocals.

As plates are characterized by being almost excessively bright, you may find yourself wanting to use a bit of high-end damping to bring it back in check. Pre-delay, as usual, can help to separate those key elements from the wash of reverb which follows in a very cool way. There are no hard and fast settings here, so just experiment until you arrive at something pleasing.

Delay

Delay is a common effect, often not heard explicitly yet appreciated subconsciously by the listener. Many engineers who find the use of reverb to be 'passé' often opt for effected delays to help create a sense of space within their mixes. Delay is interesting, as it is more of a rhythmic element than reverb, and as such can have a strong musical effect as well as an aesthetic one.

Due to its pronounced effect, delay commonly does not work well on percussive elements such as drums, unless used specifically as a musical effect. It tends to shine on vocal tracks of all types, and lead guitars in particular.

Mono delay

This type of delay is likely the most common you will find yourself dealing with. It constitutes the workhorse of your tracks, and adds a very important sense of space and tone to your lead vocal and guitar solo tracks.

The most common note values you will find yourself dealing with are 1/4th and 1/8th notes. Quarter notes often constitute very musical, pulsing delays that blend well rhythmically into the music. Eighth note delays create a bit of that 'slap back' vibe, and can be used in places where you want the effect of a delay, but without it necessarily stomping on the music.

Feedback can be set entirely to taste. Often a setting which trails the delay quickly into the distance is the best candidate, as it doesn't contain the abrupt cut-off of a simple slap-back, yet it also doesn't overstay its welcome. Keep in mind that you are using this delay on musical elements which are phrasing quickly and rapidly. You don't want the delay to get in the way of the clarity of the performance.

On vocals, a driven tube or tape type effect can be great for separating the delay from the vocal in the mix. This way you can add texture to the vocal, as well as creating more clarity for the performance, because the tone of the delay is perceptibly different, and hence clashes less.

Lead guitars have their own effects which can work well. Usually anything which thins or darkens the tone, all the while modulating it away into the distance tends to work great for texturing the performance. The key is to experiment and find what works for each individual mix.

Filtering can also be handy in separating the delay from the dry track. If you find that you want the delay to be loud, yet it is excessively masking the performance, try dialing a high cut filter and sweep it down until the obstruction is gone. Likewise, a high-pass can clean up some of the excess junk in the delayed signal and keep the core performance more audible.

Stereo delay

The first things that pop to mind when using stereo delay are backing vocals and panned lead guitars. That being said, stereo delay can also be used to effect on mono elements, either as a delay throw to keep key vocal or solo phrases hanging, or as a way to create a greater sense of space for certain mix elements.

The most common note values for stereo delays are often 1/4th notes and 1/2nd notes. Quarter notes often create a good rhythmic basis for backing vocals and panned leads. Second notes can be great for specific delay throws. Experiment with this by 'throwing' a track with a key phrase you want highlighted into the delay during the moment of interest. Throwing delays is one of the more musical aspects to mix engineering, and it can be a good bit of fun as well as adding more depth to the song. Stereo delays can often be used with darker, almost lo-fi colorations. This keeps the bulk of the performance clear, while creating a spatial density on the sides of the mix. It also allows you to turn the delays up much louder than you would be able to otherwise, which helps in creating highly energetic choruses. Filtering applies in much the same way as it does for mono delays.

Ping pong & rhythmic delay

When we get into the art of using a delay as a musical element, then just about anything is possible. Many modern delays have LFOs built in, and can be sequenced to play out just about any rhythmic pattern one can imagine. There are no hard and fast rules or settings here, as it is a creative endeavor. Just be mindful of what you're doing, check with the band if it's anything particularly crazy, and the sky's the limit.

Chorus

Chorus might surprise you in the scope of its application on a record. When used subtly, it can often enhance the texture of a mix element and make the presentation seem more vibrant. When used at extremes, it can create some interesting textures and effects.

Subtle widening

One of the best, mix-specific day to day uses of chorus is as another layer of psycho-acoustic widening and texture enhancement. The key elements which come to mind as benefiting from this treatment are backing vocals and lead guitars.

What you want to do is set the chorus to work across a shallow range, modulating very slowly. This means a low 'Rate' and 'Depth', with 'Feedback' to taste. It goes without saying that it should be a stereo chorus. Feed the relevant tracks to it gradually, until you notice the ebbing in the texture, as the phase and time variant effects of the chorus play out with the dry performance. Use this effect subtly, as when you master it, it will become one of the keys to that unparalleled, professional radio rock 'wall of vocal' sound.

Texture enhancement

Much of the tonal use of chorus gets committed on during the tracking phase of a record. For instance, chorused clean guitar sections benefit from having the chorus run right into the amp chain, as it tends to sound a lot more cohesive than being applied retroactively in the mix. Likewise with heavily chorused synths, the patches are normally committed on before being presented for mixing.

Our job as mix engineers is to use our tools for the subtle enhancement of mix elements, more so than utter sonic annihilation. As a result, there is not much to be said here beyond simply dialing the tool to the task at hand when the need presents itself. Assuming you're crafting a last-minute amp sim clean guitar sound to fit your mix, or chorusing a synth retroactively, you simply need to dial the tool to the best of your ability, with all parameters presented. As always, there are no rules, so do what needs to be done to achieve the best result under the circumstances.

Stereo doubling

Stereo doubling is one of those double-edged swords that can subtly enhance the presence of a static, mono mix element and take it to the periphery of the speakers, yet can also cause discomfort to many listeners if overdone. At its most basic level, it works by copying the signal twice, panning the copies diametrically, time delaying each by a different amount, then pitch shifting one up and the other down. This creates a faux stereo effect which can be a nice quick fix when looking to add some width to an isolated mix element.

Lead vocals, guitar solos and synths can all benefit from this treatment. Fortunately, there is not much room to go wrong with these effects, as their parameters are very basic and often work very well when left at their factory defaults. The only thing you need to concern yourself with is 'how much?'. The answer, (painfully) as always, depends entirely on the mix you are creating. My advice would be to not push the effect far past the point of audibility on something sustained and obvious like a lead vocal. If, however, you're dealing with something innately unnatural sounding like a synth, at times cranking the effect right up to the ceiling can be just the ticket in thickening it out.

Stereo widening

Now we're stepping into the dangerous stuff. There are some tools out there which can catch the ear of new comers and sound much more applicable than they actually are. Stereo wideners often fall into this category. After all, who wouldn't want their mix to sound wider? If the mix you're working on sounds somehow congested and centered, obviously a tool named the stereo widener would fix that? Well, yes and no.

We've established quite thoroughly by now that the basis of a well-balanced mix is achieved through working the basics well. 90% of it is simply good use of compression and equalization. By the time a mix has been balanced well with the basic tools, it often sounds well spread out and wide already.

The stereo widener as a tool is there for those occasions where you want to push it a little bit further into the extreme. When panning hard to either side just isn't quite enough. Within a mix these can often find utility on elements which are intended to surround the mix. Namely tracks such as rhythm guitars, acoustic guitars, backing vocals, overheads and whatever else serves similar roles within the song.

Simply because the tools can often do outlandish things to the track and create all sorts of phase-y, psycho-acoustic wackiness, doesn't mean you should use them to those extremes. Often stereo widening applied in very minute amounts can create the sort of

sound you seek, without the drawback of messing with the listener's head too much.

Remember, stereo wideners work on psycho-acoustic tricks in order to achieve their result. This isn't the same as allowing your pan knob to go past 100. It's using phase and time delays in order to fool the mind into perceiving the sound from 'beyond' the speakers. Used for excessively long periods of time, and set to extremes can often fatigue a listener or leave them feeling somewhat uncomfortable. The novelty effect you gain from using these tools to such extremes never tends to outweigh the long-term negatives of diminishing the accessibility of your work. As ever, use with caution.

Distortion

This brings us to a Pandora's Box of possibility. Subtle distortions are a large part of what make recorded music sound euphonic to us. Many of our favorite classic equalizers and compressors create substantial distortion, and it is part of what contributes to their charm and application on our work. As such, becoming familiar with the effect and application of distortion during mixing is a key element to being a successful mix engineer.

As a matter of standard procedure, the tracks we will find ourselves using distortion on the most often are vocals. Across a wide variety of genres, using distortion to effect on a vocal track is key to enhancing its perceived texture, and highlighting elements of the tone that we want to bring forward. When people perceive a mixed vocal sound, they often don't even realize that they are hearing a reasonable amount of distortion. They're simply hearing a breathy, textured vocal. Yet, how did it become this way? Why is the midrange so contoured and grainy? Does the vocalist really sound that breathy and coarse when you're sitting a room with them?

Distortion can at times contextually work when used on drums and synths. With drums,

one will often be raising the relative distortion level through all the compression, both in serial and parallel. If the right tools are used for mixing, adding distortion to drums isn't strictly necessary. Though if needed, it can often be just the ticket when it comes to thickening them out, and creating that dense, heavy-handed sound many are after. Often this is best when blended in parallel. With electronic synths, no such rules apply. Many sounds beg for distortion to be used as heavy-handedly as your imagination would allow. Let's run through some different sources of distortion, and their respective sonic qualities.

Recording chain distortion

As the name would infer, this is the distortion that you get from the recording chain itself. No recording chain is entirely free from distortion, no matter how well made it is. Not all are intended to be. One of the most common ways to attain distortion on a track is to slam the microphone preamp into the red, then use an output pad in order to bring the level back down to a tolerable spot. Each preamp has its own distinct sound, which quickly becomes apparent when used to such extremes. Fortunately we have access to plug-ins these days which emulate the sound of these units run into distortion, and they can be an amazing tool for us in creating a unique texture for our tracks.

Compressors also create distortion, some more than others. Often the level of distortion is proportional to the level of compression. This is not something you actively think about, yet when you choose a compressor to use on a vocal track, much of what you listen for in its character is the nuance of the distortion that it provides. Some transformer based classic compressors, such as the 1176 Rev A can create a very euphonic tone on vocals, and if driven hard enough can negate the need for further distortion down the track.

Stomp box / rack unit distortion

Ever felt like dusting off those guitar overdrive pedals and using them on your mix? Try it! Whether it be an old Tubescreamer, a DS-1 or a rack unit like the PSA-1, these can all generate highly usable distortion sounds for use within the mix. Often these tools can work amazingly well when obliterating synths into new and unique textures. A lesser-known fact is that they also work really well on guitars – who would have thought?

Amp & cabinet distortion

Yes, you can even run tracks through your guitar amps. The huge frequency shaping effect of the cabinet and microphone can lead to some very unique textures to fit your mix. This can work very well as a parallel blend into your vocal tracks, adding a ton of unique, analogue texture.

Miscellaneous distortion

There are no limits to what you can use to create distortion in the mix. Whether through a megaphone, an old transistor radio, a walky-talky or anything else you might have on hand. Distortion is there to be experimented with. While the bulk of your usable, euphonic, steady state distortion will be derived primarily from the recording and mixing chain, some circumstances can call for a more creative approach. Sky's the limit.

Phaser & Flanger

Finally, we wrap up with the non-specific effects. Phaser and Flanger are remarkably easy to discuss in a mix specific context because they are entirely contextual. There are absolutely no hard and fast rules whatsoever, as these tools are often used specifically as effects rather than aspects of the core sound of a track. For instance, you wouldn't often run them on a lead vocal track the entire way through a song.

Much like delay throws, knowing when to pull these tools out to effect can be very beneficial in creating a particular soundscape in your work. Often the old ethos of 'less is more' will ring true. After all, the less of an effect you use, the more pronounced its effect becomes when you do.

Chapter 11: Mastering the Mix

It's worth noting that any piece of gear or plug-in you insert on your master bus gets applied to *every single track in the entire project.* As such, it's worth delegating the very best of your gear toward this purpose. Avoid the morbid inclination to use ratty distortion or lo-fi plug-ins here, because it's the quickest way to destroy everything you've worked toward in a single move. There is a lot at stake at this point of the mixing chain. Try to emblaze the notion of doing *'as few moves as possible, but making the most of each one'* into your mind.

Mix into it

Whatever processing you have on your master bus, it's important that you begin your mix with it engaged from the start. You want the perspective of having it there from the moment you import the tracks to the moment you're finalizing the mix. A loss of perspective can be a bad thing, and applying processing retrospectively can do more harm than good.

Console saturation

This is one of the most 'no-brainer' elements to insert on your master bus if you're not mixing on an analogue console. Pulling up a good saturation plug-in, which has the right color for your track, can be a great way to gain some free glue & cohesion. It's very hard to go wrong here as long as you gain stage correctly. Ensure you are in the plug-in's level sweet spot and all should be well. If it's an accurate emulation or an actual console, the harder you drive it the more saturation effect your mix will gain. Consider it a dynamic 'transient & density control'.

Compression

The most common thing to use on the master bus of a modern, commercial mix as standard procedure is a compressor. By the nature of where it's sitting and what it's doing, this compressor *needs* to be darned good. If your workflow is largely digitally oriented and you're considering using a hardware piece anywhere, make sure that it's *here.* My personal recommendation is to use a snappy VCA-style compressor to retain all the transient content, glue the tracks together, and add a sense of movement to the mix, without losing fidelity.

Because the SSL 4000 series of desks were the facilitators of so many hit mixes from the 80s onward, many current top-tier engineers favor the quad bus compressor which was featured in those consoles.

Fortunately for the mere mortals that cannot afford an SSL console, some have isolated this design and built spin-off compressor models from the circuit. SSL themselves also offer the compressor as a stand-alone product.

Compression settings

If you are using a compressor of the nature described above, these are my recommended starting points to you:

> **Attack:** 30ms
> **Ratio:** 2:1 or 4:1 depending on the 'heaviness' and density of the song.
> **Release:** Auto, 100ms or 300ms, depending on the speed of the song and the 'action' you want from the compressor. Auto is the safest setting.
> **Gain Reduction:** Depends entirely on how aggressive you want to get, depending on the song and your mixing style. For softer material you may want to hover around 2dB, and for heavier material you can push it up to 8dB or beyond. 4dB is a great 'work with anything' setting. At higher levels of gain reduction, be aware that things start to get very 'interesting' and you will need to do a lot of automation to keep the song together. You should avoid doing high amounts of gain reduction unless you're using actual hardware compressors. As of writing this, plug-in compressors are still unable to compete when working at extremes.

Equalization

The one thing you want to ask yourself before doing any equalization on the master bus is, *'What type of EQ tends to go on all my tracks, as a matter of standard practice?'*

You want to avoid doing excessive EQ here, at all costs. For many people doing none at all would be fine. Others may be inclined to give their mixes a helping hand by including a high shelf, low shelf or perhaps both. This will inherently want to increase the depth of any raw track in the mix, and will mean less channel EQ. The one obvious advantage to this technique is making the most of a high quality hardware equalizer. If you have one of these, you can apply a degree of your standard high or low shelves to *all of your tracks,* instead of using potentially inferior channel equalization to get you all of the way there.

As with the compressor, this is the place to bust out the big guns. If you have an old Pultec, Sontec or GML lying around, now is the time to plug it in.

Equalization settings

Fight the urge to over-do things here at all costs. You're giving your mix a helpful boost, *not* mastering it.

> **Low-shelf:** 60Hz to 125Hz, 1 to 2dB maximum
> **High-shelf:** 10kHz to 16kHz, 1 to 2dB maximum

Limiting

A brick wall limiter, used *very* subtly, can give the track an extra sense of togetherness, as it controls errant transients. This isn't necessary by any means and can be taken care of by the mastering engineer, but used here can help shape the mix toward the direction of the ideal end product.

Rough mastering maximizer

If you're aiming for your mix to be LOUD, it helps to mix the bulk of it through a rough mastering chain. Inevitably, if you mix solely at around -18dBFS, the huge amount of headroom, amongst other factors, will push you to mix in a way that might not translate to a master that averages around -8dBFS RMS.

It's all about perspective. As great as mixing at a low level, with all the headroom in the world is, you have to concede that your clients will likely want the master somewhat smashed in this day and age. The best way to attain the transparent loudness that so many crave is to monitor the mix being 'smashed'. This way you can account for what happens and mix for it.

Not everybody will agree with this sentiment, but from an entirely logic-based perspective, it makes sense. Mixing toward your desired loudness will decrease the amount of work that the mastering engineer has to do, and in the end your mix will sound more effortless at higher levels. If you were intending to print the master averaging at -18dBFS then it would make sense to monitor your mix down there too.

Try to use the best maximizer tool you can. A primitive 'smash-everything' brick wall limiter might give you a skewed perspective, with all your transient content disappearing. Your mastering engineer will likely have some tricks up his or her sleeve that allows for more transparent loudness, so try using a maximizer that allows the mix to breathe.

Finally, be sure to *remove* your rough maximizer before sending for mastering. Include a footnote, letting the mastering engineer know what you used to mix through.

Chapter 12: Final Words

Avoid relying on any single element to carry your mix

Those of us who come from a musical background will tend to mix in favor of whatever we've grown up playing. Drummers tend to emphasize drums, guitarists emphasize guitars and bassists emphasize the low-end. As musicians we seem inherently egocentric, consequently overlooking one of the most important tenets of mix engineering:

"EVERY ELEMENT IN A MIX AFFECTS THE PERCEPTION OF ANOTHER"

How many guitarists take the time to craft a large drum or bass sound to augment their guitar tone? That tone which sounds so large and amazing by itself, or with weak, placeholder drums. How does it sound in a large, dense production with layers of vocals, leads, and intricate drum work? In most cases these 'huge' guitar tones will cloud the vocals, eat into the bass, and essentially lead that person to a horribly messy mix.

If you have ever analyzed a professional production, you would have very likely been taken aback by how thin and focused everything sounds when soloed. The idea behind mixing is to focus each element of that mix into its dominant zone, and subdue or entirely eliminate everything else. A good metal kick consists largely of a solid, tight low-end peak, and a thin high-end 'slap'. This leaves a good deal of space for everything in between, allowing the bass drums to be played as quickly as practical without destroying the clarity of the overall mix.

A rhythm guitar tone normally has its subs and 'air' frequencies filtered out. It also

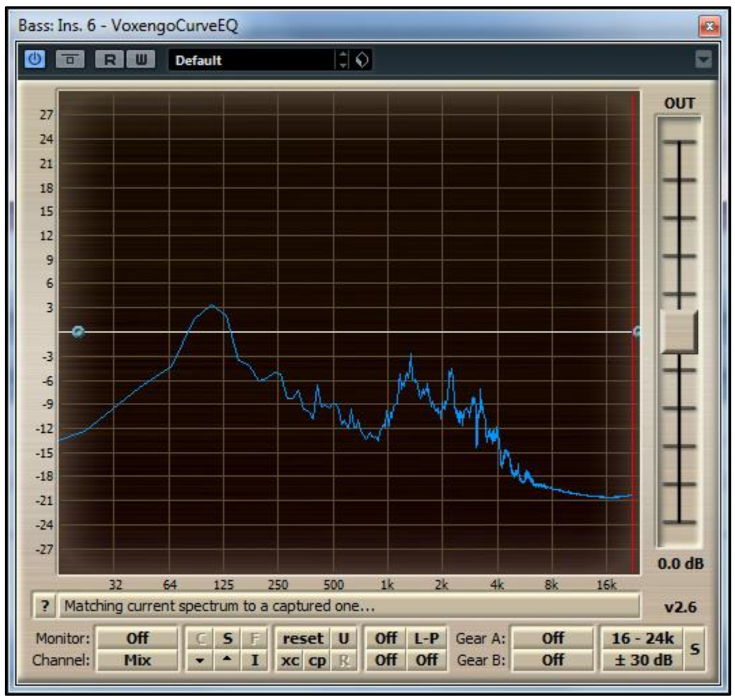
commonly has audibly unpleasant frequency peaks reduced, lows controlled and high mids in-check. This is all for the purpose of leaving room for the vocals that sit on top, and the bass guitar that sits below. There is more low-end definition when a bass guitar is allowed to do its thing unobstructed, just as a vocal is clearer when it doesn't fight for dominance with the guitars.

Every move you make is a sacrifice. A sacrifice of energy in one element to enhance clarity in another.

As mix engineers, our world is much more complex than 'turn this track up, and turn that track down''. We need to have an ear that hears frequency interactions between *all* the elements within our mix, and how they both hurt and aid each other. This way all of our tracks can be 'up' at the same time without impacting one another.

EQ matching

Contrary to the way most mixing instruction goes, this tip deals with a very effective *and* educational quick fix. It's a rare freebie.



There are a number of EQ spectrum matching tools out there, such as Voxengo's 'Curve EQ' and Izotope's 'Ozone' equalizer. The amazing thing about these tools is the potential for learning they carry. What they effectively do is analyze the spectral response of one track, and apply that spectral response to another. Now, this doesn't mean your job is as simple as analyzing your favorite raw bass track and simply applying it to your own for a 100% match - no, few things are that simple. What it means is that if you can collect stemmed versions of your favorite records, you can use them as a baseline to see how far off your own balancing is.

These tools work much more effectively on some types of tracks than they do on others. For instance, you would be best to avoid using them on vocals whatsoever. On bass guitar, however, they work amazingly, and can help with the arduous task of balancing low-end in less than ideal rooms. The way to use these tools most effectively is to get your mix as far as you can possibly take it yourself. Now, assuming that your mix resembles another, find the raw stems of either the bass or guitars. Capture the spectrum as comprehensively as you can. Apply that spectrum to your own tracks. What

you'll usually get is something very odd sounding, especially on the guitars. You then downsize the severity of the matching until the track starts sounding appropriate again. What the tool will give you is a glimpse into which areas you may want to think about boosting or cutting across the spectrum. I would suggest taking the plug-in off, then applying these boosts and cuts yourself manually using a standard EQ.

The only instrument I've found that spectrum matching can almost work on 100% is bass guitar. For some reason it takes very well to the processing, and provided that you know how to tweak the curve after it has been matched, you can actually get away with keeping the spectrum matching on there for the final mix.

Now as much of a helpful tool this is, primarily for developing one's ear and perspective, it's still a crutch, a set of training wheels, so to speak. You should endeavor to gain what you can from this technique, but after attaining enough experience and knowledge, abolish it from your workflow altogether.

You are *always* limited by the source

As you continue to work through a multitude of projects, try to bear in mind that the effort to result ratio is not always equivalent. The amount of work it takes to get one project from sounding 'great' to 'phenomenal' may only get a 'poor' project to be 'mediocre'. Always understand that you're limited by the source. There is a wide magnitude of techniques that we, as mix engineers, utilize to polish a recording, but there is still no replacement for the right ingredients at the source.

Sometimes the only way to execute a cohesive mix on a particular project is to make it sound a little messy. This doesn't necessarily imply a flaw on your end – you're simply working within the confines of what you've been given. At the end of the day, if a client requests an entirely professional mix, then that request had best have started life out well before the pre-production stage. Everything from the song content, to the arrangement, to the tempo mapping, to the overall execution affects how a mix is ultimately perceived. Your best work as a mix engineer will always sound better on a good source than it does on a poor one. That is simple reality. Grasp it, embrace it, and don't try to turn the project into something it was never meant to be.

Just throw it away

You know how the best drum techs in the world seem to have that method of tuning up all the lugs on the drum heads until they're perfectly in tune... and then just throwing one of the lugs out to get the vibe and life back into the drum? Well, the same principle applies to mixing. If you've spent weeks and crafted something resembling a technically perfect mix, the first thing you'll begin to notice is that it doesn't actually sound exciting at all. So, screw something up. Put too much low-mid in the guitar, give the kick too much low-end so it pumps the master bus compressor, make the overheads just a tiny bit harsher than they need to be, *do something wrong*.

We are imperfect beings, and we seem to look for that bit of imperfection in everything we consider dear. We generally prefer a good home cooked or chef-cooked meal to pre-processed fast food. We generally like hand drawn art over computer-generated art. We

like that one note the singer just barely missed within the stellar take, because it tells us that was a real moment in time. Likewise with your mix, you need to know what to screw up, and in what quantity in order for it to feel 'right' rather than 'boring' or 'messy'.

Push against the ceiling

A very nifty method of optimizing headroom usage within your mix is to push hard against your rough maximization chain, provided your balances are more or less in check. Try to get it up to commercial CD level. What you'll generally find at this point is one of three things:

- Everything sounds awful (the most common scenario)
- It's loud but kinda weird and pumpy (often the case if you know what you're doing)
- It sounds fantastic (sign of tweaking the mix to death, or just plain being awesome)

See, low-end is what eats into the most headroom on mixes. If your kick is bottoming out, this technique will let you know immediately. The maximizer will start pumping and tearing like mad until you get your low-end balanced. Likewise, if your high-mids are too prominent or congested then the maximizer will likely start clipping like mad - especially on guitar lead or solo sections. It's a great method to work out whether your automation is too extreme, as any elements that push outside the track too hard will immediately cause the maximizer grief. If you're after a 'perfectly' balanced mix, then use this technique in conjunction with your ears and spectrum analysis in order to ensure that mix is as efficient as possible. Ultimately this will allow the mix to translate well across a wide array of playback systems, and achieve a high overall level without as many undesirable artifacts. This means that the mastering engineer is less likely to destroy your work (even if that person is you!). Better translation, more loudness, additional peace of mind, less work for the mastering engineer. It's a win-win all round.

Always be better than your last

There are many artists in this world with differing music, differing levels of proficiency and differing financial freedoms, so it's not always possible to exceed the quality of your past projects with the ones that follow them. What you can strive to do, however, is make sure that the ratio of improvement between the song in its raw and mixed states becomes larger. As long as you come away from each project having learned something that can serve you in the future, you've won the battle. Mixing is an on-going learning experience. Keeping your mind open to improvement over time, you will continually develop your repertoire of skills to aid in realizing the goals of your clients & yourself. To that end, I sincerely hope that this mixing guide has aided you in your journey - whether you're taking your very first steps, or looking to add to an already substantial reservoir of knowledge. Ultimately the search for perfection is simply an ideal - it doesn't exist within the real world. Nonetheless it is crucial that we always strive to improve ourselves, ensuring we're ever equipped to serve both ourselves and our clients, under whichever circumstances the future may hold.